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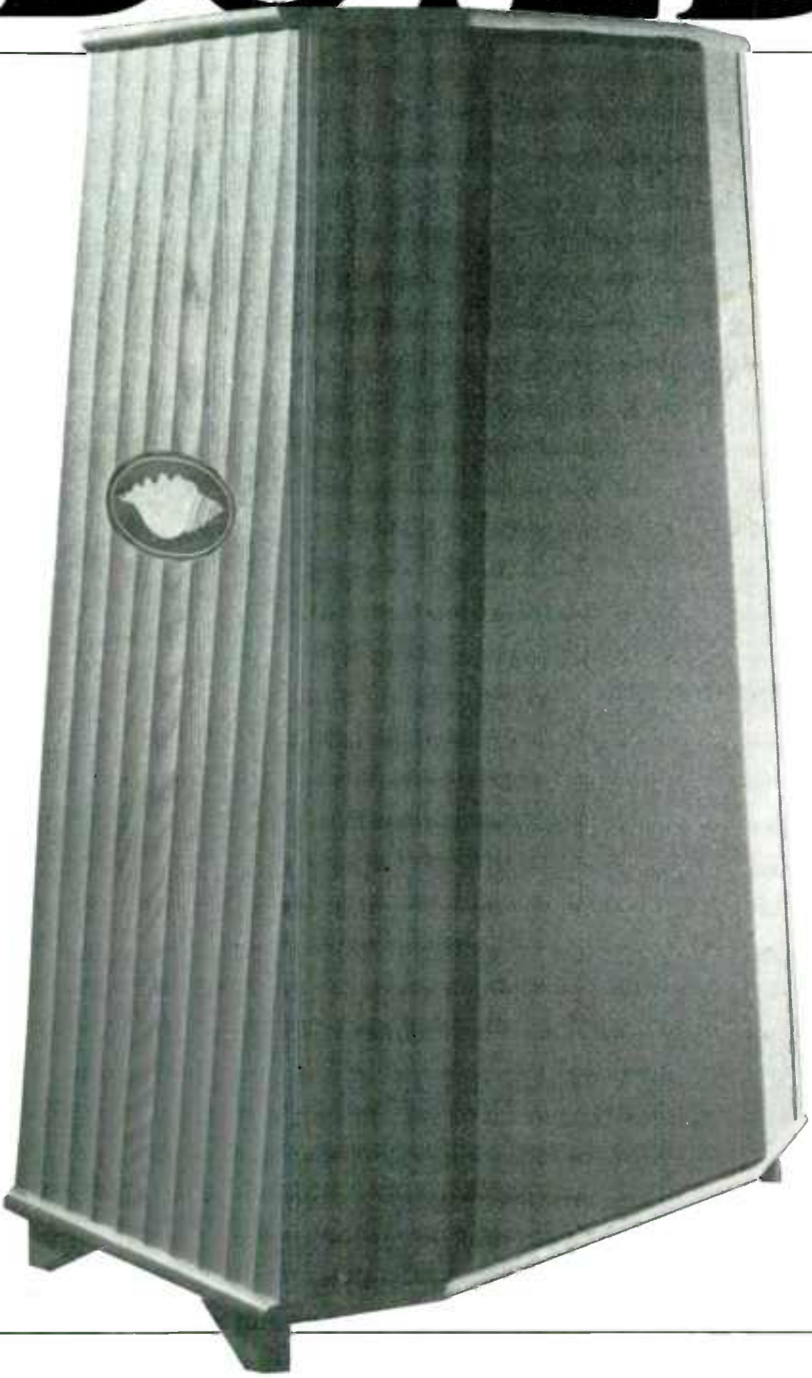


A New Driver Design

Talking With KEF's Cooke

Build A Dual Wattmeter

Good News

New from **SYMDEX AUDIO SYSTEMS** is a modular speaker system with full-range speakers and add-on woofers that combine for three-way operation and offer accurate imaging, extended frequency response, and high power-handling capability.

The Symdex speaker system features two-way Sigma speakers with diffraction-reducing, edgeless cabinetry and a movable, track-mounted tweeter adjusted to eliminate time smear and offer superior transient response for accurate imaging. A computer-designed front baffle minimizes midrange wavefront distortion by up to 9dB. The speakers provide a frequency range of 60Hz to 25kHz at ±1dB, a Q factor of 0.707 ±2 percent, and a combined power-handling capacity of 50 to 150W per channel. Passive crossover Omega woofers with "wireless" interconnection provide enhanced frequency response and power handling.

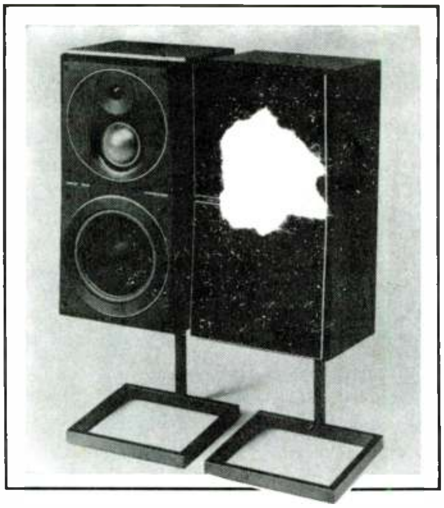
The Symdex Sigma speakers cost $900 a pair, and the Omega woofers cost $1,300 a pair. Contact Symdex Audio Systems, Dept. S33, PO Box 8037, Boston, MA 02114.

BANG & OLUFSEN OF AMERICA has introduced the Beovus S80.2 three-way loudspeaker system. Drivers consist of an 8-inch woofer, 3-inch dome midrange/Phase Link filler driver and 1-inch dome tweeter. All are suspended in rubber gaskets and mounted on a common axis.

The woofer is a new system called the "Balanced Drive," which puts the voice coil equidistant from the cone's outer edge and the top of the diaphragm. At the same time, the driver's null point is as far forward as possible to minimize phase distortion.

The midrange dome's magnet resides within the voice coil rather than outside as is customary. This change yields a rigid, distortion-free chassis and a closed magnetic circuit. The dome itself is made of a specially coated fabric that prevents breakup.

The dome tweeter is fitted with an unusually large magnet and has an aluminum voice coil. The tweeter and



midrange are bonded onto a plate, which, in turn, mounts to the baffle, yielding an airtight bond with no sharp edges to produce diffraction.

The Beovox S80.2 is capable of handling up to 80W continuous and up to 110W peak. Should excess output endanger the speaker, a protection circuit will turn off the unit before any damage occurs.

The Beovox S80.2, with a genuine rosewood veneer cabinet, has a suggested retail price of $595 per pair. For additional information, contact Bang & Olufsen of America, 1150 Feehanville Drive, Mt. Prospect, IL 60056.

The Model Four loudspeaker system is the latest offering from **SIDEREAL AKUSTIC AUDIO SYSTEMS.** It is a compact, floor-standing speaker designed to deliver flat power response in real listening environments at all audible frequencies.

For the low frequencies, an 8-inch woofer is mounted in an enclosure of acoustic suspension design. For the middle frequencies, a miniature, moving-coil unit has a total mass of 1.31 grams, with a free-air resonance below 100Hz. A flat ribbon tweeter, using a super-lightweight diaphragm of 76.5mg, reproduces the higher frequencies. An etched-aluminum voice coil produces a flat frequency response to 45kHz, with nearly 160° dispersion of all frequencies from 30Hz to 20kHz. The quasi second-order crossover design has 6dB/octave filter slopes. The crossover frequencies are 400Hz and 4kHz.

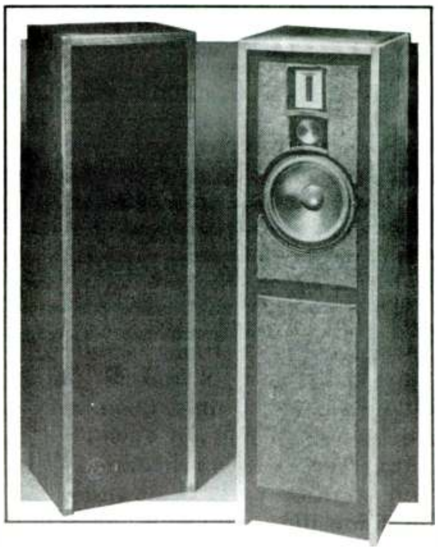
BIAMP SYSTEMS has introduced three new equalizers—the EQ/140 parametric, the EQ-220 graphic, and the EQ/230 ⅔-octave graphic.

The EQ/140 is a single-channel, four-band parametric equalizer engineered to function at low noise and distortion levels for a variety of uses. It features fully balanced outputs and inputs, ±16dB range (−40dB for feedback tuning) and five LED overload indicators.

The EQ/220 is a ten-band professional graphic equalizer with gyrator-simulated inductor circuitry and transformerless balanced line techniques to ensure low noise. It features two independent channels, ±15dB range, EQ bypass switching, overload peak indicators, magnetic field immunity and output protection.

The EQ/230 is a 30-band, ⅔-octave graphic analyzer designed for high-quality professional applications. Features include smooth, accurate filter action; a ten-segment LED ladder on each channel; tape monitoring pre/post; overload peak indicators; an in/out switch; floating and balanced input/output circuitry; a ¼-inch phone/XLR balanced/RCA plug; and 45mm slide controls.

For details on these equalizers, write to Biamp Systems, Dept. S33, PO Box 728, Beaverton, OR 97075.



For details, write to Sidereal Akustic, Dept. S33, 4035 Oceanside Blvd., Unit G57, Oceanside, CA 92054.

PYLE INDUSTRIES' two new 10-inch musical instrument sound reinforcement speakers feature 1½-inch Power-Proof® voice coils with the company's new high-temperature FRK voice-coil form. These additions to the Accent I series offer power-handling capability of 115W RMS maximum, with a 70Hz to 7.5kHz frequency response rating. The sound pressure level is 108dB.

The 8-ohm version, model number MEC10C290, costs $58.95, while the 16-ohm version, model number MEC10C290-16, costs $59.50.

For further information, write to Pyle Industries, Dept. S33, 501 Center Street, Huntington, IN 46750.

Three new music speakers from **ELECTRO-VOICE** offer increased power-handling capability and excellent low-frequency reproduction. The EVM Pro-Line Series, which is designed for professional, high-level, high-performance sound-reinforcement systems, includes several design improvements such as heat-resistant materials, low-mass edge-wound flat-wire voice coil construction and proprietary manufacturing techniques. The Pro-Line units are driven by Electro-Voice's largest 16-pound magnetic structure, while the voice coil and the magnetic structure are vented to maximize heat dissipation.

The new EVMs are available in three sizes and five models to fit virtually any design application. The 15 and 18-inch models can handle 400W continuous power (per EIA Standard RS-426A) and short-duration program peaks of up to 1,600W. The 12-inch EVM Pro-Line speakers are rated at 300W continuous with 1,200W program peaks.

The EVM-12L Pro-Line loudspeaker is intended for extended-range sound reinforcement applications, or in the case of musical instrument systems, for lead guitar-type applications. Almost identical to the EVM-12L, the EVM-12S is designed for more emphasis in the 2,000 to 3,000Hz range for added brilliance and punch in full-range uses. The EVM-15B and EVM-15L Pro-Line loudspeakers are intended for general purpose low-frequency reproduction, the latter for

systems that also require response above 3,500Hz. The EVM-18B Pro-Line offers exceptional low-frequency performance, including heavy fundamentals in the 30 to 40Hz range.

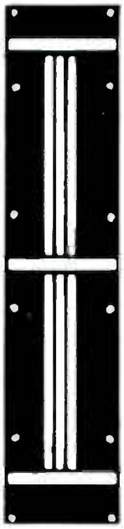
The EVM Pro-Line characteristics are appropriate for both vented (bass reflex) and horn enclosures. Electro-Voice engineers have designed six vented enclosures specifically for EVM speakers. They span low-frequency limits (3dB down) from 38 to 83Hz and internal volumes from 1.3 to 13 cubic feet. They may also be stepped down for more extended bass response, with low-frequency limits ranging from 27 to 58Hz.

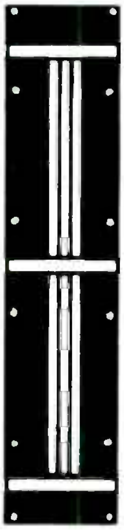
Prices of the EVM Pro-Line loudspeakers range from $240 to $395. Additional information is available from Electro-Voice, Dept. SB-4, 600 Cecil St., Buchanan, MI 49107.

A new catalog of precision tools and computer and telecommunication equipment is available from **JENSEN TOOLS INC.** The 160-page catalog contains more than 2,000 tools of interest to field engineers, technicians, computer and telecommunication service persons, and electronics hobbyists.

Major categories covered include test equipment, micro-tools, soldering equipment, tweezers, screwdrivers, cutters, drafting supplies, power tools, computer accessories and circuit board equipment. Also included are many new products from Jensen and more than 40 pages of service kit and tool cases for electronics specialists and technicians.

To obtain a free copy of the catalog, write to Jensen Tools Inc., Dept. SB-4, 7815 S. 46th Street, Phoenix, AZ 85040.





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About This Issue

Another step in the long search for the theoretically perfect driver is what **Dr. Carl Pinfold** believes he has taken at his University of Liverpool laboratories in Britain. Following the logic of E.J. Jordan's tiny, stiff, "full-range" driver, Dr. Pinfold has "stretched" the conventional cone and magnet somewhat like a rubber band. The details begin on page 7.

Bruce Edgar concludes his two-part interview with Raymond Cooke, KEF's managing director, on page 13, while **G.R. Koonce** carries on with his full workbench of lab instruments, this time unveiling a dual wattmeter on page 20. Part five of **Robert Carlberg's** odyssey (p. 28) gets into phasing issues and is followed by **Robert Kral's** tip on improved imaging (p. 30).

Extraordinary craftsmanship is evident in Canadian **John Otvos's** handiwork on our cover and in "Craftsman's Corner" (p. 32), reminding us that cabinetry can be a whole lot more than a six-sided box.

Note: If your address label's top right corner code reads XX83, it is time to renew your subscription. Don't miss 1984's meaty articles, construction projects and fresh ideas.

SPEAKER BUILDER

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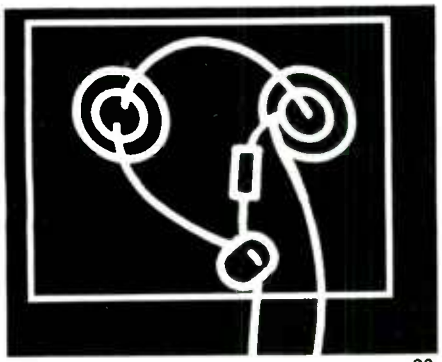
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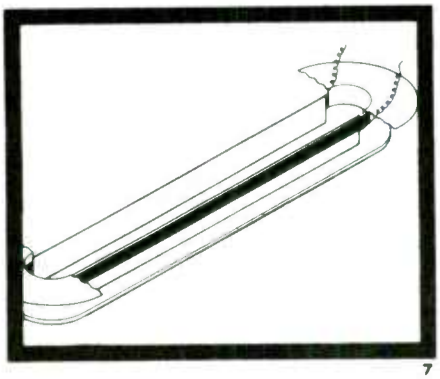
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FEATURES

DEPARTMENTS

Editorial

The Loudspeaker: Last Frontier

At the 74th Audio Engineering Society meeting in New York recently, I was struck by the amount of presentation time devoted to loudspeaker system design. Test methods were discussed, machine and subjective, and design approaches were explored—especially for large auditoriums and as applied to passive crossover networks. Of the 79 papers delivered, 25 were on some aspect of loudspeaker system design or testing.

The final transformation of electronic impulses into some kind of physical motion is still the most stubborn, baffling, intriguing and insufficiently understood problem in audio today. The battle with mother nature over this problem moves forward slowly on many fronts. We are glad to note that KEF is taking some positive steps by measuring its drivers and passive crossover components to match their characteristics to what we know about driver and crossover behavior. Contributing Editor Bruce Edgar's interview with Raymond Cooke in our last issue and in this one gives us valuable insight into some of the unsolved problems facing the final transducers in our audio reproduction chains.

KEF's Laurie Fincham delivered an invited paper on that company's impulse-testing techniques, which measure loudspeaker performance to within 1dB down to 20Hz. R. A. Greiner and Travis A. Sims presented a paper illustrating a current and velocity feedback system to reduce loudspeaker distortion. I found it interesting that they used hybrid crossover arrangements in their systems—electronic for the low end and passive for the higher frequency crossover point(s). The richness of the loudspeaker papers was particularly evident in the 74th technical meeting.

One other fact struck me forcefully about this meeting: the papers delivered—especially in the area of crossover design—were largely the work of amateurs. As those of you who have subscribed to either *Audio Amateur* or *Speaker Builder* probably know, I hold firmly to the primary meaning given in current dictionaries for the word "amateur." That meaning is a description applied to a person who does an activity for the love of it and not for money. Unfortunately, a secondary meaning is now recognized and is becoming more pervasive. More and more people are using the word to describe someone who is inept or inexpert. The prime root of the word is the Latin for "love," which clearly indicates its origin, in contrast to what I consider the misuse and ultimate threat to its usefulness.

Four of the six papers delivered in Session F ("Loudspeakers: Network Considerations") were written by amateurs. All but one of these authors are academics, and three are mathematicians. However we may wish to classify the seven men who presented the results of their work, it is worth observing that only two of them are even remotely related to the business of making loudspeakers. Their theoretical work in the areas covered is valuable information for those who manufacture loudspeakers and crossovers. Of course, we are aware that an academic who presents a paper at such a gathering usually has some important reasons for doing so that might have little to do with his love for the subject. "Publish or perish" is not an empty slogan for university teachers. But I do know from personal talks with all but one of these gentlemen that they are deeply motivated by what I would describe as an amateur's passion for good sound. They include R.M. Bullock, J. A. D'Appolito, R. A. Greiner, John Vanderkooy, Stanley Lipshitz and Eugene Zaustinsky.

Where would we be in this world without the amateurs?—E.T.D.

ONE-WAY DRIVE

BY DR. CARL PINFOLD

Fundamentally, the design of moving coil loudspeakers for the mass hi-fi market has changed little over the past 30 years. Rectangular wooden boxes of varying dimensions contain two or three drive units. These are fed through chokes and capacitors so that each handles a discrete portion of the frequency spectrum. Generally speaking, variations on the theme have been fairly minor (some would say subtle) and have consisted largely of substitution of one material for another, and of small dimensional changes, some for sound engineering reasons, some for commercial expediency.

So much expertise and industrial know-how has been lavished on such loudspeaker systems that to break with tradition and go along a different track is a hazardous venture and is probably why radical alternatives are scarce. With one or two exceptions (notably Lowther and Jordan), the development of moving coil drive units to cover the whole audio frequency spectrum was abandoned some 30 years ago. The last serious attempt was, perhaps, by E.J. Jordan, whose metal cone 50mm "module" arguably deserves better from the British hi-fi public than it has received (although report has it that it is highly regarded elsewhere). But the present almost universal use of direct-coupled amplifiers makes the use of a single-drive unit without a crossover an especially attractive proposition. It enables the voice coil to be directly connected to an amplifier in a way that preserves high electrical damping.

The most common commercially available loudspeakers derived from the "monitor" approach—or claiming such pedigree—use two drive units crossing over at 2.5 to 3.5kHz. It is worth noting that this is the frequency band where the ear is most sensitive and most discriminating. Such systems require a cone diameter of 100 to 200mm to handle wavelengths whose dimensions are comparable to its diameter. Associated tweeters with no more than a $5cm^2$ diaphragm area are then required to operate in frequency regions where music often contains its greatest power levels, and thus the crossover problem is further compounded. Where additional midrange units are employed, these are often little different from the drive units to be found in car radios, television sets and portable radios.

An alternative approach to the ubiquitous design philosophy just described is to begin by designing a good middle unit, optimized to cover the frequency range that contains the most crucial musical and spatial (stereophonic) information. Then you must work at extending its performance in the upper and lower octaves.

Jordan, in designing his metal cone 50mm unit, proceeded from the fact that in a cone speaker, as frequency rises, an ever decreasing area of the cone close to the coil continues to radiate. His objective was to design a cone that smoothly changed from being relatively large at low frequencies to being effectively small at high frequencies. This he achieved by carefully sizing and shaping it and by using a material in which sound travels fast, thus raising the frequency at which sound energy at the inner and outer areas of the cone are in opposite phase to one another. It is this phase lag between inner and outer areas of a cone that causes it to bend and buckle, producing irregularities in its response and attendant coloration.

Electrostatic speakers are as good as they are partly because they do not suffer from this phenomenon—all parts of their diaphragms are set in motion in unison by an equally distributed force. But in a conventional moving coil speaker, a 1-inch-diameter motor coil at the apex of a cone is expected to control the movement of a relatively large radiating area. Cones, being rather flimsy structures, do not behave in the piston-like fashion that is expected of them.

GREATER CONTROL. The crucial dimension in the issue we are discussing is that between the edge of the coil and the outer edge of the diaphragm. On this dimension depends the degree to which the whole of the diaphragm is controlled by the coil. Jordan went for a controlled non-piston-like movement, still using a relatively small coil and large cone. Another approach, which is adopted in this new drive unit, is to reduce the distance between the coil and the diaphragm extremity to achieve more closely piston-like movement over most of the audible frequency range. Briefly, this involves grossly enlarging the coil in relation to the size of the diaphragm.

Figures 1A and *1B* show two ways in which this can be done. In the configuration of *Fig. 1A*, a 10cm x 10cm flat diaphragm of "squared circle" shape is driven by a voice coil of 7.5cm diameter. More than 60% of the diaphragm area is within 1.5cm of the coil's edge. In the configuration shown in *Fig. 1B*, where an elongated diaphragm 20cm long x 4cm wide is driven by a coil 19cm long and 1.5cm wide, no part of the surface is further than 1cm from the elongated coil. The control exerted by the coil over the surface of the diaphragm in both these configurations greatly exceeds that exerted by the conventional coil over its cone. The time lag between the energy leaving the coil former and reaching the most remote parts of the radiating surface is much reduced, so that the latter moves almost entirely in phase up to quite high frequencies.

Configuration B has certain further advantages. First, because the drive force is widely distributed, all the bending, buckling, crushing and sheer forces acting on the diaphragm are greatly reduced. Although, being flat, it does not have the advantage of conical shape to lend it stiffness, longitudinally it is stiffened by the ribs that hold the coil. Additionally, transverse stiffness over the small span involved is readily provided by the material of which it is constructed. On the other hand, being flat, it does not suffer from the megaphone-like colorations of conventional cones whose frontal volumes form resonant acoustic cavities. Secondly, for a sharp stereo image, the sound source must be substantially narrower than the distance between human ears. This applies particularly to reproduction at middle frequencies, where much of the time-delay spatial information in stereo material resides.

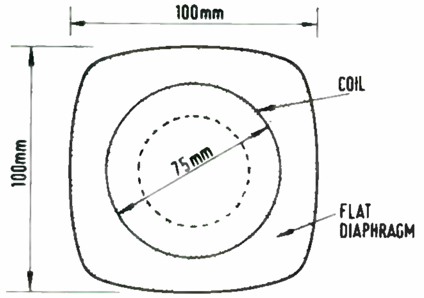

FIGURE 1A: The dotted area, which constitutes 60% of the diaphragm area, is within 15mm of the coil edge.

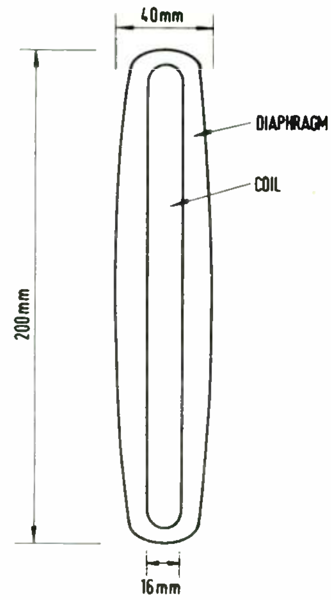

FIGURE 1B: No part of the diaphragm is more than 12mm from the coil edge.

LIMITATIONS. There are two obvious limitations to this combination of large coil and relatively small diaphragm area. A small diaphragm, unless it is to make impractically large excursions, cannot be expected to produce extremely high acoustic power levels, particularly at very low frequencies. Moderately loud undistorted reproduction is possible in reasonable sized domestic rooms, but the power levels expected by some heavy rock enthusiasts are beyond the system's capabilities unless a subwoofer is used.

The second limitation is the well-known fact that small speakers are insensitive. Even with the substantial magnets used in the prototypes, it has not been possible to raise the sensitivity above 86dB for 1W at 1m on axis. But this sensitivity, which is typical of "monitor" loudspeakers, simply means that amplifiers in excess of 40W are required to obtain acceptable loudness. This is by no means unusual for small speakers, and, indeed, most modern high-quality amplifiers provide power of this order.

Units of the kind shown in *Fig. 1A* have also been developed, and while stereo imaging is somewhat less focused, they have proved rather more sensitive than configuration B units. Some listeners marginally prefer the slightly different tonal quality they produce. This is attributable to a rather more pronounced mid-frequency region. One or two engineering and production problems on these "A" units still await resolution, so we will initially market a unit based on configuration B.

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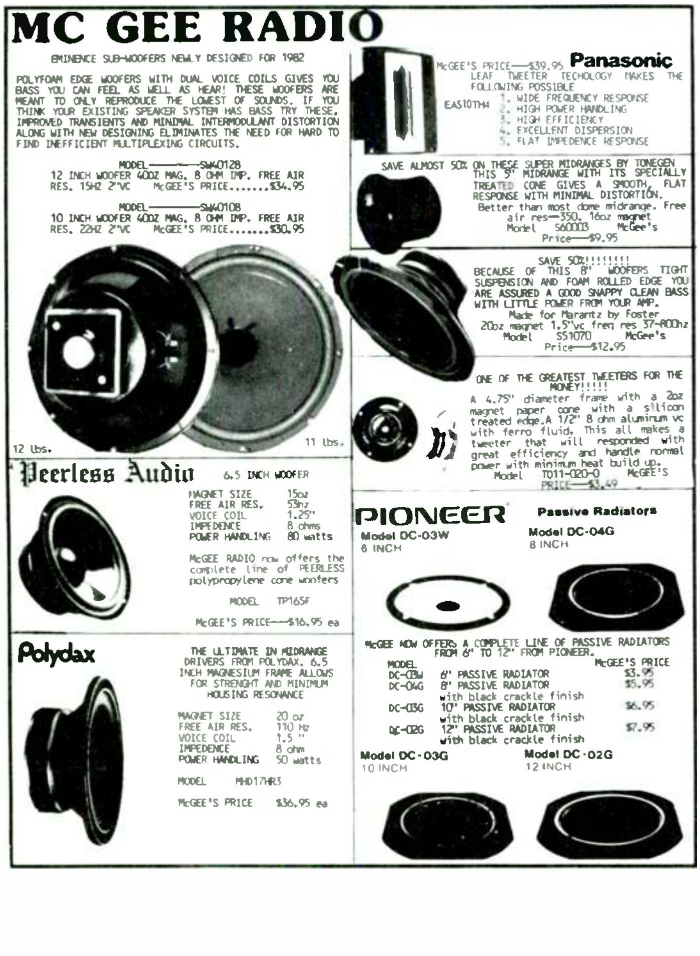

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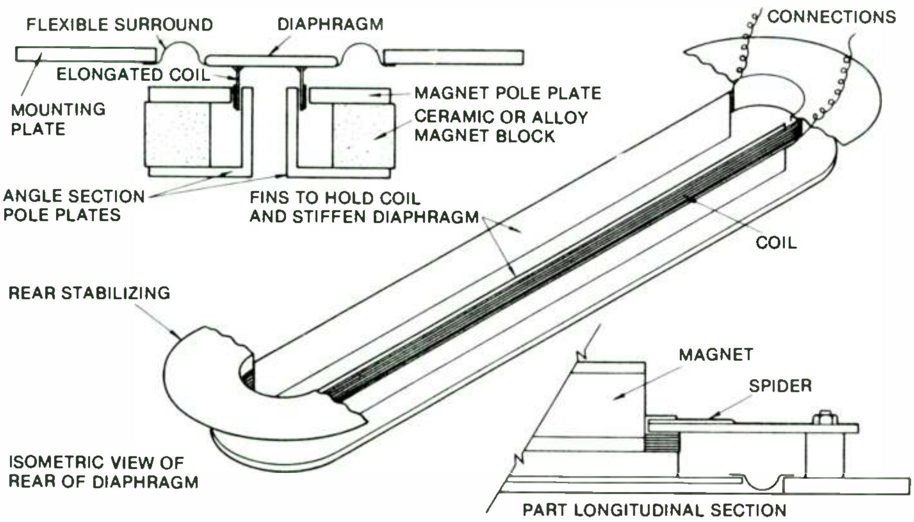


FIGURE 2: Various views of the Pinfold diaphragm.

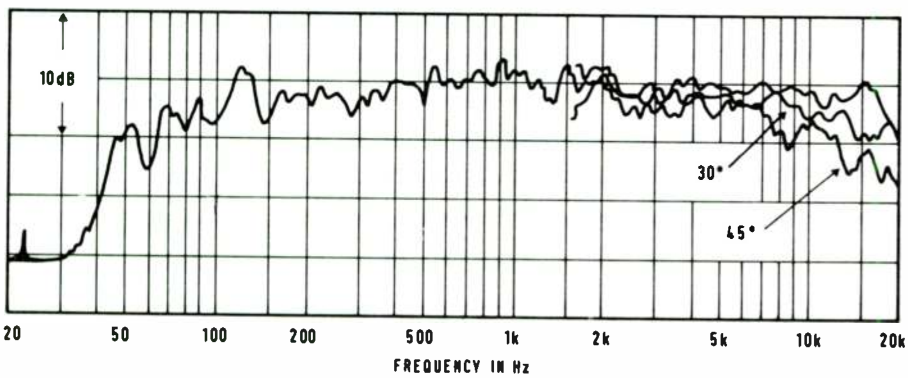


FIGURE 3: Measured responses, both on and off-axis, of the speaker using a new diaphragm constructed of a combination of materials.

CONSTRUCTION. Now I'd like to describe the construction of the diaphragms for the two configurations and the enclosures in which these drive units can be housed most advantageously. Choice of materials for the diaphragms is actually rather limited. They must be stiff, light, and possess a degree of self-damping, so that when they flex—as inevitably they must to some degree—they do so uncatastrophically. Over a long development period, diaphragms were constructed by laminating thin sheets of balsa wood, the grain of each sheet being angled to the others. As sound travels much faster along the grain than across it, these laminated structures possessed a welcome degree of self-damping. The surfaces of a balsa diaphragm have to be sealed off from the atmosphere to make it dimensionally stable with changing humidity. This was done with a thin layer of self-adhesive plastic film.

These balsa diaphragms behave very nicely right across the frequency range and produce little coloration, although their construction is relatively labor intensive. Organic materials, however, are unpopular in industrial applications because of the difficulties of quality control, so more recently we have developed a lightweight laminated plastic diaphragm that behaves as well as the balsa one and has none of the latter's production problems.* The rest of the construction details are best gleaned from the accompanying photographs and drawings.

There is little point in designing a drive unit that is free of coloration and then housing it in a resonant enclosure. Enclosures add far more coloration to loudspeakers than is usually recognized. They are, after all, rather like large areas of sounding board coupled to the drivers, in much the same way the sounding board of a piano is coupled to its strings. Large areas of timber serve a welcome purpose in grand pianos, but are less helpful in loudspeaker design. Fortunately, small drive units flourish in small enclosures that are made up of relatively small, and hence rigid, panels.

For the new drive units we have developed fairly small enclosures constructed of ceramic tiles on frameworks of cork. Each tile is mechanically separated from its neighbors by cork and mastic. The rigidity of the ceramic and the system of decoupling the panels, as well as the decoupling of the drive unit itself, seems to produce an acoustically "dead" system, which contributes to the general neutrality of sound.

These units have taken a long time to develop because every slight change in dimension or in the physical properties of different components and materials has an effect on frequency response. Whereas in a multiple-drive-unit loudspeaker aberrations in each drive unit can be pushed out of its operating range and frequency balance can be adjusted in the crossover network, in a single-unit loudspeaker such latitude is not available to the designer.

Loudspeaker design has become very much the province of specialized professionals who have a wide range of materials, processes and technologies at their disposal, along with the elaborate modern test gear needed to monitor results. Some of these professionals, notably at

*Since this article appeared in *Hi-Fi News & Record Review*, the suppliers of the material for Mr. Pinfold's original plastic diaphragm have ceased manufacture of that product. In a letter to *Hi-Fi News & Record Review* (June 1983, p. 19), Mr. Pinfold writes, "We finally settled on a combination of materials, which is more stable and a little easier to construct. As a bonus, the new structure has resolved some torsion mode problems which were adversely affecting the response." See *Fig. 3* for the measured responses, both on and off-axis, of the speaker using the new diaphragm.

Wharfedale and Celestion, have been enormously generous with materials, technical know-how and the use of measuring equipment without which this development would not have been possible. Among those who have helped are the late G.A. Briggs, Ken Russell and Gareth Milward of Wharfedale; Colin Aldridge, David Inman and Graham Bank of Celestion. Tony Kennedy and Mike Armstrong of Preformations and Burt Hudson of Magnet Developments have always seen that the right magnets were available.

As a teacher of architectural acoustics at the Liverpool University School of Architecture, I have enjoyed the use of a laboratory and measuring equipment as well as a teaching schedule that allows some time, mostly during vacations, to pursue a research interest. Naturally, the University holds the patent rights. □

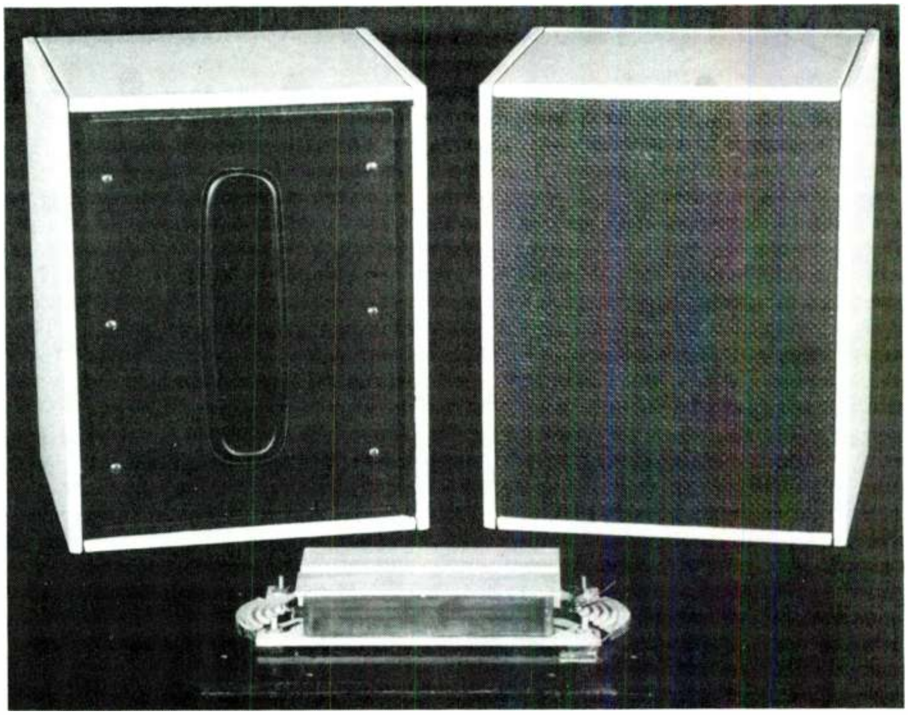


FIGURE 4: Mr. Pinfold and his associates now sell this boxed unit using his new driver.

AN INTERVIEW WITH RAYMOND COOKE: PART II

BY BRUCE EDGAR
Contributing Editor

In Part I of this series, Raymond Cooke described his career before he founded KEF, his first product at KEF (the K1 slimline) and his early design philosophy.

SB: Aren't you getting into the problems of stereo imaging and all its dependent factors?
RC: Yes. I became aware of this in the early '60s and late '50s when we got into stereo reproduction. It seemed evident to me that some speakers gave a better stereo image than others, but there were no means of quantitatively testing for it. One problem with loudspeakers is that unless you are listening in an anechoic chamber, you cannot hear what the loudspeaker alone is doing. You are always listening to a combination of direct and diffuse sound, which has come from nearby reflective surfaces.

The best we could do was to take a good speech recording (i.e., one that sounded natural, but that had a known "pedigree") and listen to it in a room that was as acoustically symmetrical as we could find. (See *Fig. 4.*) We would check to see if the voiçe appeared to be coherently in one place or if it appeared to move about with the spectral content.

I remember worrying about the problems of stereo imaging, trying to quantify it and, more importantly, find out what caused its breakup. We simply noted that some speakers gave good stereo images and others didn't. In addition, some monitors gave better stereo images than others.

FIGURE 4: KEF auditions its speakers (in this case, a pair of Model 105s) in a specially designed listening room. Sound absorbers on the walls and ceiling help to control reflections and produce conditions for a good stereo image.

SB: Was this difference in stereo image between the same or different monitor models?
RC: The difference was between different loudspeaker designs, not monitors of the same design. We knew that this had something to do with frequency response and directional response—what I call distribution, not dispersion.

Everyone uses the word dispersion to describe distribution, the way high-frequency sound is spread. The difference between distribution and dispersion is as follows: the old lady bathing in the ocean and peeing at the same time is dispersion because the rate of dilution is such that at a distance away from the source, the effect of the piddle is negligible. Distribution is the old gentleman, who has had a bit too much beer to drink, peeing over the balcony rail in a Victorian music hall and spraying it about a bit.

SB: How did you come to the conclusion that stereo image had something to do with distribution and frequency response?
RC: I think the real daylight came about ten years ago at an exhibition in Paris. It was the first exhibition put on in the new Palais de Congres, where they hold the Festival du Son. Our distributor there had taken a large demonstration room

and stacked all our loudspeaker products at one end in an aesthetically appealing way.

I was working the exhibition with Laurie Fincham, our technical director. As I was listening to one floor-standing model, which had been selling quite well, I said, "I think that loudspeaker sounds awful." He agreed with me. The loudspeaker was at the bottom of what the distributor called a loudspeaker wall. I didn't know the cause and was tempted to blame the records, but that wasn't the case. When we played a speech record, it was obvious that something was wrong. I didn't know whether it was the influence of the other loudspeakers, a bad sample, the room acoustics or the amplifier. So we arranged to have the loudspeaker shipped back to the factory.

The speaker had been designed using our own drivers and the old methods of "cut and try" in designing the crossover network. In that method you try to produce a network that divides the spectrum on a terminal volts basis. At the same time you try to present a feasible load to the amplifier that does not become wildly reactive or goes down to very low impedance. It should also give a plausible response curve and sound right. You simply start off with a basic network and tinker with the values and components until you satisfy the four conflicting requirements.

When the speaker was back at the factory, Laurie did a thorough investigation that showed the sample was entirely within our manufacturing tolerances. He did, however, notice a large difference in sound quality when listening on the principal axis and at other angles in the vertical plane. Placing the loudspeaker high in the room produced one effect, while placing it on the floor produced another—not only at low frequencies, but also at the middle and high frequencies. We were able to attribute this change in balance to large variations in the midband frequency response at different angles.

We found a disparity between the character of the sound perceived on the listening axis and that perceived on the measuring axis, where people conventionally measure response. Also, the principal acoustic axis, which may be different from the listening and measuring axes, is highly variable in the crossover region because the polar distribution pattern is altered by the phase relationships of the contributing drivers.

This situation is well understood in radio frequency terms. Anyone who has worked with antennas knows that you can swing the polar response of an antenna by phase retarding or advancing one of the antenna elements.

SB: Why didn't it occur to you or others that the same thing would happen in acoustic terms?

RC: I don't know. We did understand the elements of it in a way. I remember one BBC monitoring loudspeaker we made for many years. It was the LS5/1A,[7] with two high-frequency drivers, one placed above the other in vertical array. The two units provided sufficient power-handling capacity in the midband. The system included a 15-inch paper cone unit and ¾-inch high-frequency units. The 15-inch unit had to struggle to get up to the high frequencies and was used well beyond the safe region up to 1,500Hz. On the other hand, 1,500Hz was mighty low for the tweeters to handle. They would have been happier coming in an octave above, but by stacking them it was just possible to get down to 1,500Hz to meet the woofer on its way out. This posed a problem—you couldn't place two tweeters close enough to operate in the many kHz region. At 10kHz or more, you simply obtained a bunch of bananas for the distribution pattern.

The idea was to roll off one of the high-frequency units by phase retarding it. It was understood that if you phase retard one driver, the polar pattern in the crossover region is tilted downward. We elected to tilt it downward because most studios have carpeting, which is a fairly innocuous reflecting surface. An upward tilt would give reflections off the ceiling.

SB: How did you change your design procedures to incorporate the new ideas about phase?

RC: We decided that the listening axis was the goal and that we should organize all the phase responses to drive the speaker in the preferred way along the listening axis. (See *Fig. 5.*)

SB: What kind of crossovers do you use?

RC: That depends on what we are trying to do. The configuration of the network comes out of the system specification. The "musicality brigade" cling to their belief in simple first-order networks. But these cannot work properly because in most cases the 6dB/octave network requires electronic boosting of the output outside of the passband. Therefore, the 6dB/octave roll-off can be used only in electronic or active crossovers.

The normal 12dB/octave roll-off cannot work well because the phase relationships in the crossover region are always a problem. You have to go to third or fourth-order filters to obviate some of the undesirable side effects, such as the main lobe of the distribution flopping about in the crossover region.

You should not consider the crossover in isolation; you have to consider the drive unit, the box and the filter together. We do it by the "target function" approach. (See *Fig. 6.*) It invites you to say, "What do I want from this loudspeaker,

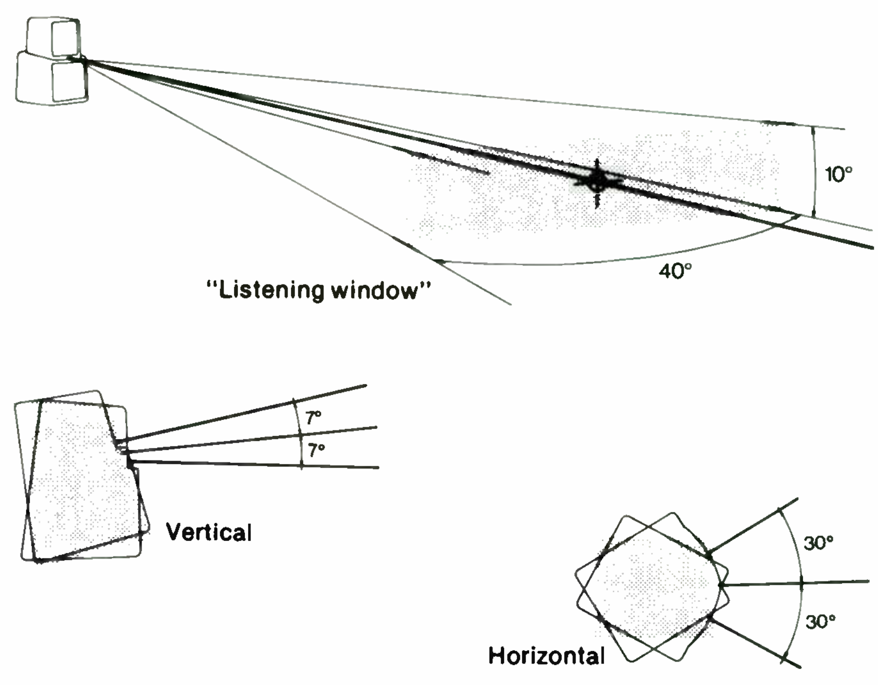


FIGURE 5: This illustrates the range of adjustments of the listening window for the Model 105. This is a standard KEF specification.

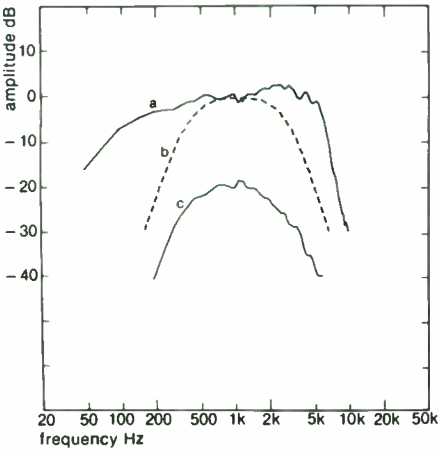


FIGURE 6: KEF uses the target function to determine the joint effects of the drive unit, the box and the filter. Curve (a) shows the measured frequency response of a midrange driver without a filter. Curve (b) represents the desired response of the filter and driver together. Curve (c) shows the response of the filter that achieves the target function. The curve is displaced to show its shape.

and what is its overall frequency response?" Then you take the drive unit and place it in the box geometry you intend using. You measure the amplitude and phase response of the unit in its enclosure geometry and subtract the measured response from the target function. You finish with a difference curve in amplitude and phase, which says what the network has to contribute. You have to find a network that will bridge the gap between the raw driver in the box and the target function.

SB: Is this difference curve hard to fit with a realistic network?

RC: Generally speaking, it is readily synthesized. Occasionally, you do get a bridging function that requires some extra elements to deal with a dip or peak that results from something in the box geometry. The secret in driver production is to be able to reproduce the driver consistently, idiosyncrasies and all. Once you've designed a network to deal with those idiosyncrasies, they have to stay at the same amplitude, phase and frequency so that your crossover responds in the same way.

SB: What do you do about the response peaks that invariably appear at the top end of the driver response?

RC: We try to get rid of the peaks by treating the driver first. One loudspeaker design on the English market has a screaming peak in the high-frequency end at just over 20kHz. The designers attempted to deal with this by putting a suck, or notch, circuit into the crossover filter. The only problem is that the Q of this high-frequency peak is dependent on various quirks in the manufacture that are hard to control.

One problem is that the filter element has to be tuned by winding turns off of an inductor. The danger in this is twofold. First, it relies on the drive unit retaining its peculiarities throughout the life of the speaker, and second, if you have to replace the drive unit, the replacement will probably not match the original. Another problem is that the suck circuit goes to very low impedance at high frequencies, so at about 35kHz the network wanders down to an ohm or so. If there is any signal up there, it will trip the protection circuits of the amplifier.

After making sure that all the design work conforms to the intended listening axis, you must make sure that the frequency response from both loudspeakers is the same. We match the responses with a computer selection program. Having produced the world's first identically matched pair of speakers, we were astonished how much this improved the stereo image.

SB: How did KEF get involved with the BBC monitor loudspeakers?
RC: It all started back in the '70s when the BBC was experimenting with its matrix H quadraphonic transmission program and wanted four loudspeakers that were as closely matched as possible to 0.25dB over the midband. We had the gear to do this because we were just getting into computerized measurement. The BBC actually used them in a listening room at Royal Albert Hall during a Prom season. They were obviously pleased with the results. It gave us the impetus to do matching on a wider scale. Of course, now we do it as an everyday feature in sets of two.

SB: How do you match your loudspeakers?
RC: We measure and grade all the drive units, then have a computer search through a batch of 94 to find the proper mate in terms of sensitivity and frequency response. If we can't find a mate for a unit, it is dropped off. (See *Figs. 7, 8* and *9.*)

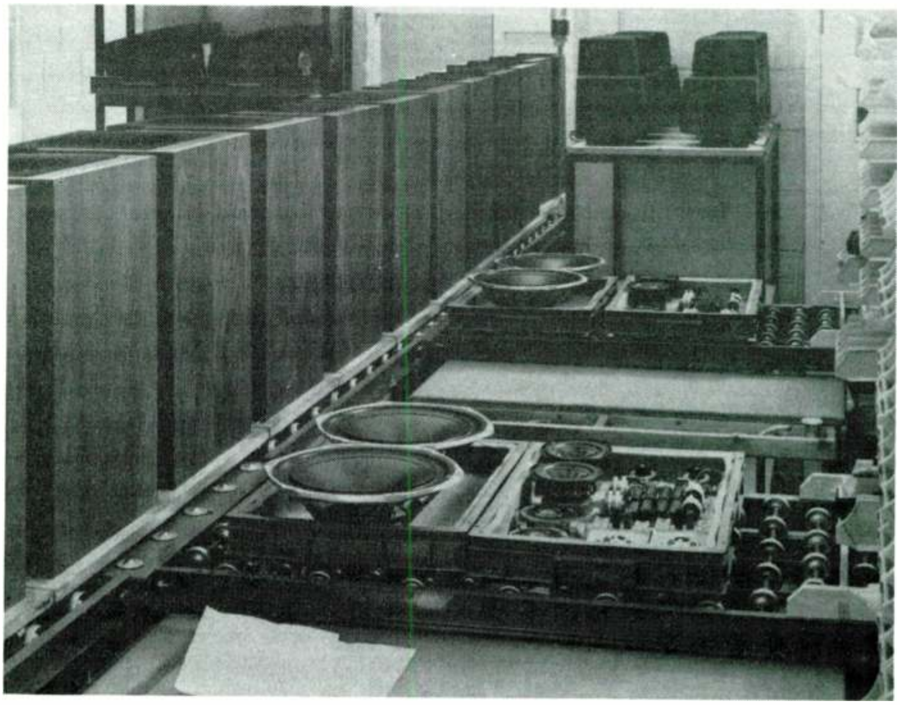


FIGURE 8: Here matched sets of drive units and crossover networks are sent to the assembly line.

FIGURE 7: To find the proper mate for each drive unit, KEF uses impulse response techniques to measure each potential mate in a 94-unit batch.

SB: What are the criteria for a good match?
RC: It is normally within 0.5dB. We can set that limit anywhere, but for domestic uses, 0.5dB is reasonable. For every pair of woofers we have to find complementary midranges and tweeters. We then have dividing networks, which are closely matched by tolerancing. Finally, we measure the whole system to make sure that the selection process has worked and nothing has changed.

SB: How do you choose the tolerances for your crossover components?
RC: It is far too expensive to choose by tolerancing, so we buy capacitors and resistors in bulk and make our own inductors. They are all measured and graded in 2 percent tolerance batches, so there might be a 15 percent spread for capacitors. For every network we have a routine that tells us that if one component strays in one direction, we can balance that change by moving another component in a different direction. A complex chart tells us how to use components a long way from the design center. Some components are critical, and some are noncritical. (See *Figs. 10A* and *10B.*)

SB: What kind of capacitors are you using now?
RC: For the large ones we use nonpolarized electrolytics. We use a particular make of capacitor with which we have worked for 15 years. There is a lot of nonsense talk in the US and Japan about the nature of the dielectric. Paul Voigt used to say, "The poor little electron neither knows nor cares what force is pushing him." The people who don't understand these things at an engineering level have the notion that electrons are like swimmers in a vat of oil: the electrons can tell the difference between the fluids.

Because these people don't understand things such as loss factor, they jump to conclusions that Mylars are better than electrolytics. They change capacitors, listen to the results and claim that any differences are due to the dielectric. They ignore the fact that the loss factor and all the strays are different. They don't know that a capacitor isn't a capacitor—it is a conglomeration of linear and nonlinear resistances and capacitances—so they are reading only the effects of stray capacitances.

FIGURE 9: These Model 105s are awaiting final acoustic and listening tests before being shipped out.

Audio is riddled with this sort of thinking. Some people take one piece of equip-

KP-4C SUPER CLEAN HEADPHONE AMP

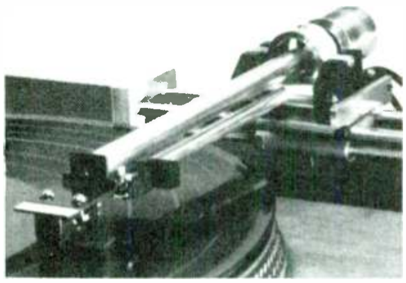
Two Channels for only $128!

- Designed by Jim Boak
- Can be optimized for either electrostatic or dynamic phones
- Direct coupled, using a 540 op amp
- Class A operation
- Low noise and distortion
- High slew rate
- Extended overload capacity gives you a clean, wide open tube-like sound
- Well regulated power supply, independent for each channel

The headphone output on most amps usually is a large resistor in series with the output to knock the level down to what is suitable for driving headphones. The resistor reduces the amp's damping ability, leaving your headphone drivers less tightly controlled. You will also suffer the distortions produced by the resistor's particular nonlinearities. To solve these problems, Jim Boak has designed a high performance amp specifically for headphones. Built around the powerful 540 op amp, it features a circuit modification, derived from Walt Jung's work, which gives the op amp a highly linear transfer characteristic. The resulting distortion curve has that gentle rise associated with good tube amps, and a clean, wide open sound quality you will enjoy thoroughly every time you listen on headphones. To top it off, Boak gave it excellent power supply regulation. Each channel has independent power supplies using three-terminal regulators to deliver stable voltages across the entire audio range. The amp may be configured for driving either electrostatic or dynamic headphones, and parts for both are included.

KP-9 TANGENTIAL TRACKING TONE ARM

Only $160!



Here's your opportunity to own a tangential tracking tone arm—and have the fun of building it yourself. Rod Cooper's ingeniously simple design elegantly achieves the benefits long attributed to tangential arms. The kit has been sold in England for several years and has been praised for its high quality. The metal parts are precision-machined for a solid, tight-fitting assembly. Mount it on your choice of turntables, and you will have an outstanding piece of equipment which you can use and enjoy every day, for many years of listening enjoyment.

- Improved tracking ability lowers distortion
- Maintains tangency $\pm 0.2°$
- Low mass (10g) tracking arm of rigid Dural
- Servo controlled, with opto-electronic feedback
- Low friction gimbal suspension

SBK-C2 PHASE-ALIGNED ELECTRONIC CROSSOVER

Two channels for only $134

One of the problems introduced by crossovers is the phase delay they cause. Over the years, many attempts to minimize this problem have been made. Bob Ballard's solution starts with a simple method of measuring the drivers' phase relationships in the air. Then he shows you how to build a crossover with phase adjusting circuits and tells you how to put your system in alignment, from the preamp outputs to the sound in the air.

- Three-way active crossover
- Phase shifting circuits to fine tune your system
- Easy method to measure actual acoustic phase relationships among drivers
- Op amp filters simplify design
- Rumble filter built in

METAL FILM RESISTORS

10¢ each in quantities less than 100, 8¢ each for 100 or more

Old Colony is closing out its entire stock of Draloric metal film resistors. Take advantage of this opportunity to stock up at bargain prices. These precision audio grade resistors are made by Telefunken to exacting standards. They have tight specs at a high level of performance with long term reliability and stability—all to assure you of high fidelity in the audio equipment you build.

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  - less than 1 microV/V to 100k
  - less than 25 microV/V to 1 Meg
- Temperature Coefficient: 50 ppm/C°
- Tolerance: less than 1%
- Maximum Power: 0.35W @ 70°C derated linearly to 0W @ 165°C
- Maximum Voltage: 250V
- Copper leads coated with eutectic solder

All standard E24 values are available except: 47.5, 121, 274, 2.21k, 3.92k, 4.75k, 8.25k, 12.1k, 18.2k, 22.1k, 24.3k, 27.4k, 39.2k, 43.2k, 47.5k, 56.2k, 82.5k, 150k, and 825k.

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Value ______ Qty. ______ | Value ______ Qty. ______ | Value ______ Qty. ______

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Phone Orders: (603) 924-6371 (MC or Visa only) 9-4 EST

For full kit listing, send stamped, self-addressed envelope to: Old Colony Sound, PO Box 243, Peterborough, NH 03458 ment out of a chain, substitute X and say, "Since I hear a difference, it must be due to X." They don't realize that they have altered the interface condition, and this produces a different frequency response, which is what they hear.

SB: So you find the electrolytic capacitor quite stable for your purposes?

RC: Yes. You have to make sure that you are getting a high-quality capacitor whose loss factor is right for the circuit. We buy capacitors on the basis of capacitance, loss factor and tolerance.

SB: Are the electrolytics you use available to home constructors?

RC: Several firms in England specialize in components for constructors. These firms, such as Falcon, Badger and Wilmslow, buy from the same source as we do.

SB: Is there anything we have not covered that the average speaker builder should know?

FIGURE 10A: Capacitors used in crossover networks are measured individually and sorted into 1 percent tolerance batches.

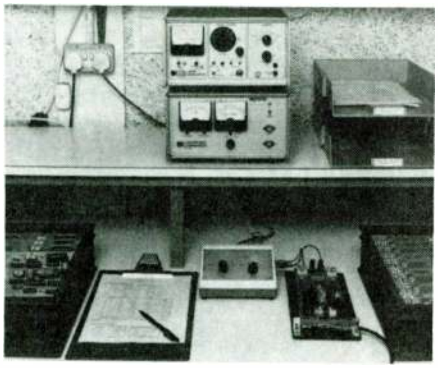

FIGURE 10B: The crossover networks are checked to within ±0.5dB of a reference standard and are constructed in pairs to difference limits of 0.2dB.

RC: I'd like to address three concerns. First, the speaker builder should appreciate the specific nature of the crossover network and the box geometry. To do that, you have to take a look at where the business came from. Twenty-five years ago almost the whole loudspeaker business was in components. In the days when ElectroVoice held sway with its range of drivers and speaker kits, it was common for the amateur constructor, even the most erudite, to take a tweeter from one manufacturer, a midrange from another and a woofer from yet another. Regardless of any impedances or parameters, he would marry all the drivers with a crossover from a fourth source and then put them into a cabinet of dubious geometry and dimensions. Finally, he would pontificate about its performance. Now we know what a load of nonsense that was because crossover networks have to be specific to a group of drivers.

The crossover network, moreover, has to be specific to a drive unit in a specific geometry. If you substantially alter the box geometry so that you change the distance from the box edge to the drive unit, the response of the loudspeaker will vary significantly. You must take this into account in arriving at a target function. Once you have settled on the type of driver and box geometry, the only thing you can alter to achieve the target function is the network.

People who are working on a certain set of drivers and box geometry write to us and ask for a schematic of a crossover that has been designed for quite a different purpose. They don't understand the significance of the network and that it is electrically and acoustically specific to a speaker design. So the first item to convey to speaker builders is that crossover networks are speaker specific.

SB: What is the second point?

RC: We need to supply drive units in matched pairs because even in the best regulated families, drive units tend to differ significantly. In uncontrolled production, the variations in sensitivity and response are enormous. I've seen variations in sensitivity of as much as 6dB in drive units that purport to be the same thing. Even in cases where the differences in sensitivity are not as drastic, you still get quite significant variations in actual frequency response. Just as in the days of push-pull amps it became necessary to supply tubes in matched pairs, it is now necessary to think in terms of matched pairs of drive units, at least for the more esoteric system.

SB: What is the third item?

RC: It is important for the amateur speaker builder to think about protection. Unless you are going to use a loudspeaker well below the safe rating, protection is an essential feature of a modern loudspeaker—if the price will stand it. (See *Fig. 11.*)

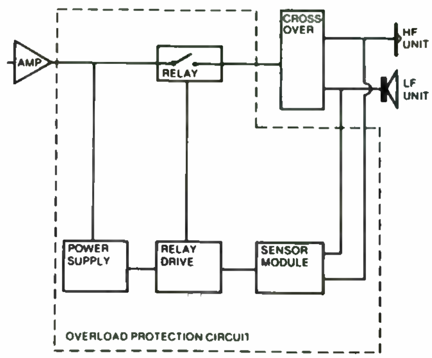

FIGURE 11: KEF uses this overload protection circuit in the Model 101.

SB: What do you recommend for protection circuits?

RC: The only way is for manufacturers to supply the essential circuitry. Here again, some Japanese manufacturers got it wrong a few years ago by coming onto the market with omnibus protection devices that didn't work because the manufacturers hadn't studied the way a loudspeaker fails—i.e., the thermal overload. We have studied thermal overload, made special test instruments, and determined in what circumstances and by what mechanisms loudspeakers actually fail in service.

SB: How do they fail?

RC: They fail mainly by thermal overload. This is a process where the rate of power input to the voice coil exceeds the dissipation rate or is sufficiently long term to override the natural thermal capacity of the voice coil, particularly in tweeters. We give the thermal capacity of the voice coil on all our data sheets. It is more or less directly related to the mass of copper in the voice coil. In a large voice coil, the thermal capacity is large enough to absorb the transient of heat resulting from a fairly heavy musical passage or from an accidental overload, so long as it isn't a steady-state signal.

In high-frequency units, however, the voice coil can't be massive, or it would not be a tweeter any longer. So we have to take that into account and put in a circuit that enables the speaker to differentiate between false and musical signals. Generally speaking, the simplest high-frequency unit is able to operate on musical signals with inputs up to a quarter of a kilowatt. Yet that same tweeter can be destroyed in a few seconds by 10W.

Many soldering irons achieve the 300 to 400°F necessary to melt solder with an input of 10 to 15W. That means that the steady-state dissipation of 10 to 15W in a confined space in a component of small thermal capacity will rapidly raise its temperature to the melting point of solder. Of course, you are in trouble with a voice coil long before you actually destroy it. Most people think that you burn out the voice coil. That isn't so. The problem with thermal overload is that you either bring about irreversible distortions in the voice coil (e.g., create an oval shape) or cause a blister due to outgassing of volatile material.

Even though a voice coil is not destroyed, the adhesive bonds are damaged. As the voice coil heats up, its resistance rises, and as the resistance rises, the current through the voice coil is lowered, reducing the mechanical force from the magnet on the voice coil. Also, the impedance shifts, as can the crossover point. These things occur at temperatures well below those that will permanently damage the voice coil.

SB: Where do you see loudspeakers going in the future?

RC: I think there are two distinct avenues for progress. First, we can further improve standards of production to ensure greater consistency in frequency response. Second, we can substantially reduce the nonlinear distortion products, which are at least a factor of ten greater than those in any other part of the audio chain. Once we get around the problem of mechanical transduction at the input end, get rid of the pickup and the analog tape recorder, and get much cleaner digital signals, I think the high distortion levels present in dynamic loudspeakers will remain a stumbling block.

SB: Do you think the moving coil loudspeaker still has a future?

RC: Yes, partly because a lot of mileage is left in the technology and also because there is no serious alternative.

SB: So you don't believe that electrostatics and other types of technologies are going to take over?

RC: No. That is not to say anything bad against electrostatics. We are merely facing up to the fact that electrostatics, by their physical limitations (i.e., ionization and physical size), are limited in what they can achieve. Even today no electrostatic loudspeaker on the market can achieve the sound pressure that the majority of the customers demand.

Two other things affect the future of loudspeakers. The first challenge is to make loudspeakers less obtrusive. Now speakers are large, ugly boxes that ruin the decor and get in the way. Many utilitarian devices of the past—such as a Louis XIV commode or an old clock—serve as room decorations long after their usefulness is gone. When the analog loudspeaker passes from the scene—as it eventually must—I can't imagine that anyone would want to keep one around just because it's so lovely to look at.

Second, user convenience ultimately gets into the act in the development of every consumer product. When you develop a motor car that can go fast enough, can accelerate well and is quiet enough, you can improve other features such as safety, visibility and acceptability. This is something that has hardly been touched in loudspeakers. For instance, we must solve a whole host of problems with the interface between the room and the loudspeaker. Work has already started on using microprocessors to enable loudspeakers to deal with this interface. We can use microprocessors and digital filtering to sort out some otherwise intractable problems. You will see more of that work, especially from KEF. □

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BUILD YOUR OWN WATTMETER

PART II IN A SERIES ON MODULAR TEST INSTRUMENTS

BY G. R. KOONCE
Contributing Editor

In Part I of this series, the author described a voltmeter (VM) circuit board and several power supply boards. In Part II, he will explain how to build a dual-channel wattmeter for speaker testing, using the boards from Part I.

When making acoustic measurements on speaker enclosures or individual drivers with continuous signals, you must have some means of determining the drive level to the unit under test. This level is usually referred to as "how many watts" you are delivering to the unit, but with speakers this terminology is technically incorrect. Speaker systems have complex impedances (i.e., they are not only resistive) that change with the frequency and the acoustic environment. Measuring "true watts" into a speaker is, therefore, difficult. Some equipment measures true watts,[1] but in general, speaker tests are defined in terms of a voltage level that would deliver so many watts into a fixed resistive load.

For example, to find the sound pressure level (SPL) at 1 meter for a 1W input, you do not drive the speaker with 1 "real" watt. You drive it with a voltage that would deliver 1W into a resistor that has the same value as the nominal impedance (R) of the speaker system under test ($V^2/R = W$). Thus, a nominal 8Ω system would be driven with 2.83V RMS, which would translate to 1W (i.e., $(2.83)^2/8 = 1$).

My wattmeter is a two-channel, average V^2/R meter, where R is switch selectable at 4, 6, 8, 12 or 16Ω for each channel. The full-scale range of each channel is selectable as 1, 10 or 100W, but you could extend it. Overload protection is provided for each meter movement, along with an overload indicator to show when readings are invalid.

Each channel also has a switchable time constant on the averager. The fast-mode time constant (approximately 0.17 sec.) is quick and convenient for adjusting continuous wave (CW) signal levels during testing, while the slow-mode time constant (approximately 1.9 sec.) is better for monitoring program material or noise signals. When monitoring program material in the slow mode, the meter changes slowly and gives some indication of how hard you are driving a system. In the fast mode, the needle kicks upward with sounds such as drum strikes, but this unit does not indicate wattage peaks and is not intended as a substitute for volume unit (VU) meters in setting record levels.

TABLE II-1

RMS SINUSOIDAL INPUT VOLTAGES

Average Power (Watts)	Value of Nominal Resistance R_N 4Ω	6Ω	8Ω	12Ω	16Ω
½	1.414	1.732	2.0	2.45	2.83
1	2.0	2.45	2.83	3.46	4.0
2	2.83	3.46	4.0	4.90	5.66
5	4.47	5.48	6.32	7.75	8.94
10	6.32	7.75	8.94	10.95	12.65
30	10.95	13.42	15.49	18.97	21.91
50	14.14	17.32	20.0	24.5	28.28
100	20.0	24.5	28.28	34.64	40.0

From $V = \sqrt{P \times R_N}$; P in watts, V in volts RMS

Note that these meters are calibrated to show the wattage developed by sinusoidal inputs. Since they use full-wave averaging, such meters will read low on noise signals. For true Gaussian noise signals (white, pink or otherwise), you should raise the meter reading 1.03dB or multiply the power reading by 1.27. With noise input, a meter reading of 4 is actually closer to 5 (V^2/Ω). If you plan to test speakers primarily with noise signals, you could calibrate this 1.27 factor (for power) right in.

Table II-1 lists the RMS sinusoidal voltages corresponding to various wattages with different nominal resistances. The case requiring maximum sensitivity is 1W full scale, with nominal resistance (R_N) equal to 4Ω. Under these conditions, the RMS voltage input is 2V. In Part I of this series, I presented a VM requiring 4V RMS for full scale. Rather than add a times-2 amplifier, I simply upped the averager gain by two on the VM board. Therefore, a 4V DC averager output is still full scale, but only a 2V RMS sinusoid input is needed to produce it. This affects the averager resistor and capacitor values. See *Table II-4* for revised VM construction information.

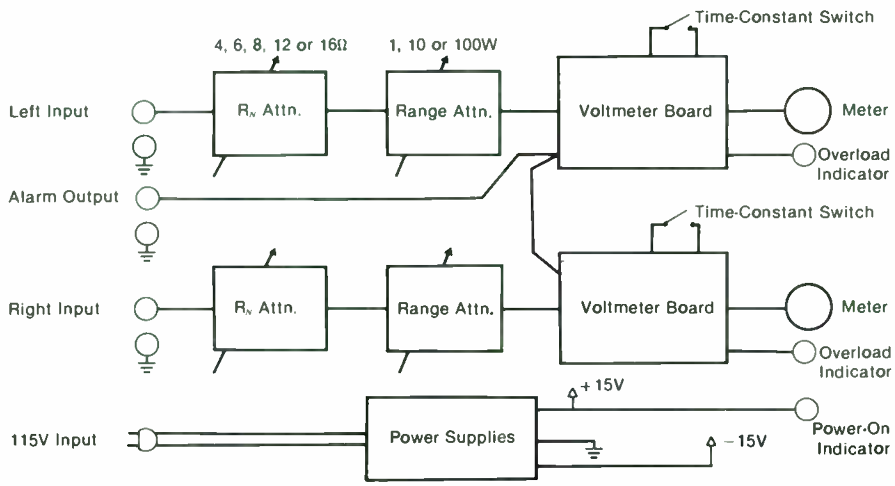


FIGURE II-1: Block diagram of the dual-channel wattmeter.

OVERALL FUNCTION. *Figure II-1* shows a block diagram of the dual-channel wattmeter. The first attenuator for each channel allows you to select the nominal speaker impedance (R_N) for which the unit will read V^2/R_N. This attenuator displays an input impedance of 300 to 600Ω. This low input impedance has the disadvantage of taking some power from the amplifier driving the speaker, but it is nearly immune to hum and other pickup problems.

The first attenuator output drives a second attenuator, which selects the full-scale meter range. In turn, the output of the second attenuator drives the VM circuit board, which means the VM board input can be severely overdriven if you set these attenuators incorrectly or apply high power inputs. The VM board has no specific protection against this, but in two years, using 80W amplifiers, I have not hurt anything. If you test with kilowatt amplifiers, you should consider a protected buffer, which isolates the VM boards and possibly the input attenuators from any setup mistakes.

Once you have corrected the input voltage for the desired value of R_N and selected the full-scale range, all that remains is to perform the "voltage squared" function. As with most inexpensive wattmeters, you do this on the meter scale. *Figure II-2* shows the scales I made for the meters in my wattmeter. Once again, don't try to scale any values from *Fig. II-2*, as my meters do not have linear deflection angle versus current. *Table II-2* gives the percent of full-scale deflection for linear movements at various powers for 10W full-scale meters. You can convert these values to degrees of deflection once you establish the 0-to-full-scale deflection angle for your meter and its linearity.

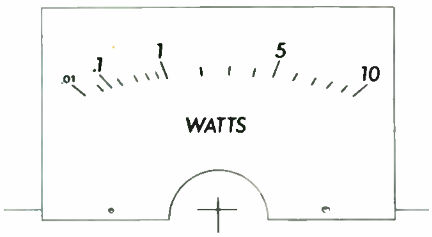


FIGURE II-2: The author cautions about scaling values from his wattmeter scale because his meters do not have linear deflection angle versus current.

TABLE II-2

WATTAGE VS. PERCENT METER DEFLECTION

Watts	% Deflection	Voltage at TP-2 on VM Board
10	100 (full scale)	4V DC
9	94.87	3.795
8	89.44	3.578
7	83.67	3.347
6	77.46	3.098
5	70.71	2.828
4	63.25	2.530
3	54.77	2.191
2	44.72	1.789
1	31.62	1.265
0.8	28.28	1.131
0.6	24.49	0.980
0.4	20.00	0.800
0.2	14.14	0.566
0.1	10.00	0.400
0.05	7.07	0.283
0.01	3.16	0.127

Note: For scale design of linear meter, full scale equals 10.

Table II-3 includes the information you need to construct the wattmeter power supplies. Note the references to the correct figures in Part I. The AC wiring is shown schematically in *Fig. I-10*, which also lists the required parts.

Table II-4 provides the information you need to build the wattmeter's VM boards. Again, this table shows the revisions in the parts list and refers to the correct drawings in Part I of this series. The value shown for R24 is for sinusoidal signals. If you decide to calibrate your wattmeter for Gaussian noise, change R24 to 43kΩ. Using resistor values other than those listed will affect the averager's resistors (R24 and P1). When P1 is finally adjusted, you find that:

$(R24 + P1) \cong 2.22(R2 + R3)$ for sinusoids.

$(R24 + P1) \cong 2.5(R2 + R3)$ for Gaussian noise.

This will allow you to select proper values for R24 and P1 if (R2 + R3) is *not* equal to 19.25kΩ.

I selected the resistors for the two input attenuators in each channel from my junk box, so I cannot give a specific parts list. I used low power resistors to trim the power resistors to the correct values. *Figure II-3* shows the information you need to select resistors for these attenuators and gives the final value needed for each resistor, along with the voltage across each resistor in normal operation and under "worst case" incorrectly set attenuators. The worst case considered is 100W into 16Ω, or 40V RMS input. If you anticipate more serious "mistakes," you might want higher wattage resistors. In designing these attenuators, I ignored the VM board loading (approximately 9.43kΩ). The worst case error will occur at 16Ω R_N and 1W full scale and is about 1.5 percent

TABLE II-3

INFORMATION ON WATTMETER POWER SUPPLIES

Special Parts

- + 15V power supply (7815 type; design for 88mA)
  R6—56Ω, 6.5W (actual max. dissipation = 0.52W)
  C8—10μF, 25V electrolytic (17V)
  Heatsink—Thermalloy Inc. 6025
  Other parts—Table I-6
  Parts size list—Table I-7
  Schematic—Fig. I-14a
  Circuit board—No. 249Z, negative in Fig. I-17
  Board parts layout—Fig. I-15a
- − 15V power supply (79L15 type; design for 15mA)
  R2—330Ω, 1W (actual max. dissipation = 0.15W)
  C5—10μF, 25V electrolytic (17V)
  R3—6.8k, ¼W (0.042W)
  Other parts—Table I-4
  Parts size list—Table I-5
  Schematic—Fig. I-12b
  Circuit board—No. 246Y, negative in Fig. I-5
  Board parts layout—Fig. I-13b

(0.14dB). I did not consider it worth correcting.

Figure II-4 shows the overall interconnection diagram for the circuit boards of the dual-channel wattmeter. Input to each channel is via a screw-type barrier-terminal strip. The right channel uses a three-terminal strip for input, return and chassis ground. The left channel uses a five-terminal strip to allow "alarm" outputs (see Part I and *Fig. I-4*) from the two VM boards, which are tied together, along with the return. You can wire this output and the other equipment to a central audio or visual alarm to warn of overloads. Although this protects meter movements, you might want to add an alarm to alert you of stress on the input attenuators.

Note that the two input channel returns tie the two "low side" outputs of your test amplifier(s) together. This can produce hum or even fireworks in your system if your amplifier has unusual output wiring such as that on a Carver M-400. Putting a resistor in each return lead does not work here because of the low, variable input impedance. If you have problems, tie both wattmeter returns to the same amplifier low side and put up with a slight wattmeter error in one channel.

The wattmeter is immune to signal phase, so driving one channel, inverted, into it causes no problem. My test setup uses two independent amplifiers, and the ground loop resulting from the wattmeter hookup produces no hum. If the ground loop problem really bothers you, you can use two AC power transformers and two of each power supply to make the two channels totally independent. This requires that each VM board's alarm output have its own return and that you keep these returns independent.

Part I covers the requirements for the LEDs and the meter for the VM boards, as well as the parts for the AC portion of the power supplies. I used SPST miniature toggle switches for the time-constant switches, but you can use anything you have on hand.

CONSTRUCTION. Before going into the construction details, I want to mention that my cabinet was the most expensive part of the unit because I selected it to fit in a particular spot. The cabinet is a miniature aluminum console consisting of a KK4P-ND top panel and a KK4C-ND case (available from Digi-Key Corp., P.O. Box 677, Thief River Falls, MN 56701). It has textured finishes on both halves and would not take rub-on transfers directly, so I had to label the front panel controls by mounting aluminum sheets with the controls. I

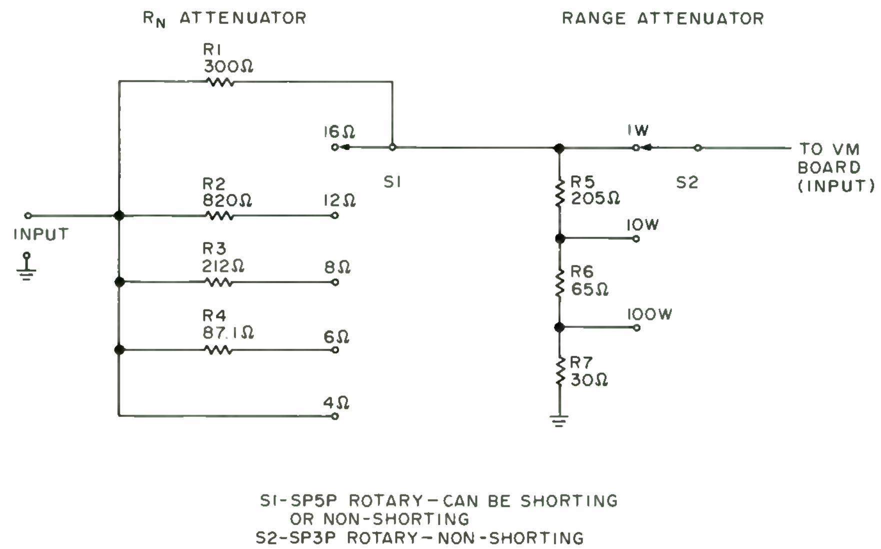
TABLE II-4

INFORMATION ON VM CIRCUIT BOARDS FOR WATTMETER

Revised Parts List

R12—15k, ¼W (≅0)
R24—39k, ¼W (7.9mW)
P1—10k (1.2mW)*
C5—4μF, 150V tantalum, foil (17V forward, 1V reverse)
C6—40μF, 25V tantalum, foil (17V forward, 1V reverse)
All other parts—Table I-2
Parts size list—Table I-3
Schematic—Fig. I-4
Parts layout—Fig. I-6
Circuit board—No. 246, negative in Fig. I-5

*See Fig. I-1L.

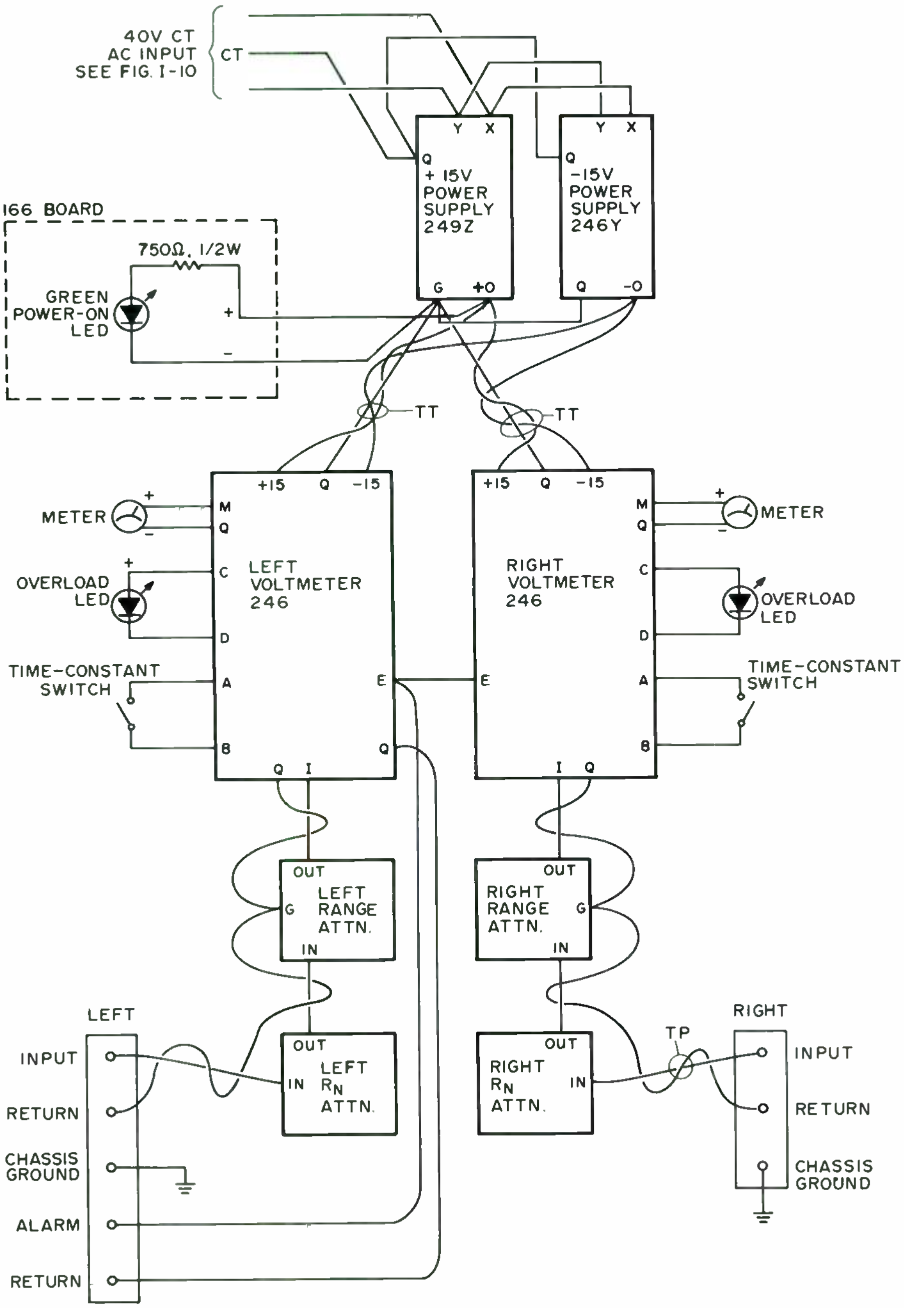


Resistor	Value	Worst Case[1] Voltage	Worst Case[1] Wattage[2]	Normal Full Scale Voltage	Normal Full Scale Wattage[2]
R1	300Ω	20V	1.33W	20V	1.33W
R2	820Ω	16.9V	0.35W	14.6V	0.26W
R3	212Ω	11.7V	0.65W	8.3V	0.32W
R4	87.1Ω	7.4V	0.62W	4.5V	0.23W
R5	205Ω	27.3V	3.64W	13.7V	0.91W
R6	65Ω	8.7V	1.16W	4.33V	0.29W
R7	30Ω	4.0V	0.53W	2.0V	0.13W

[1]Based on 100W into 16Ω (40V RMS) input with R_N attenuator in any position.

[2]Into a single resistor used to obtain the value.

FIGURE II-3: This information should help you in selecting resistors for the input attenuators.



Note: No connection to chassis by any of this circuitry. You can connect "Return" to "Chassis Ground" at terminal strips.

FIGURE II-4: Overall circuit board interconnections for the wattmeter.

FIGURE II-5: Because the author chose his cabinet to fit into a special spot, it was the most expensive part of his unit.

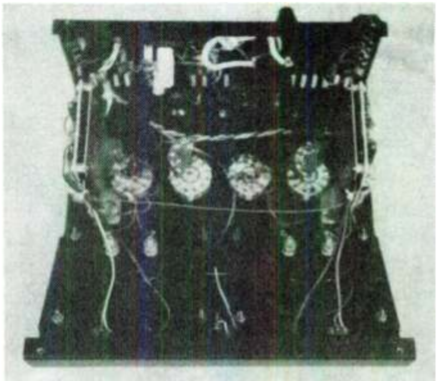


FIGURE II-6: This internal view of the unit shows off some of the author's handiwork.

painted the sheets blue and put white transfers on them. To label the back panel connections, I used black transfers on scraps of circuit board (copper stripped), which I mounted under the terminal strips. *Figure II-5* shows the outside of the unit.

See *Fig. II-6* for a look at the inside of the wattmeter. I have not provided guidelines such as drilling templates, but I have included drawings and a detailed description of each part. *Figure II-7* shows the placement of the various parts and the basic AC wiring. In this figure, the console top panel is in an open, or flat, position. Note that in the description "right" means the unit's right and thus appears on your left in this drawing. The line cord and fuse are on the rear, with one insulated post provided and the metal oxide varistor (MOV) wired from the fuse holder to this post.

The power switch and transformer are mounted on top of the unit, while the two power supply boards are mounted on the rear wall via ¼-inch-outside-diameter by ¼-inch-long 4-40 threaded spacers and ½-inch-long 4-40 screws. The power supplies are wired to the VM boards via individual twisted triplet wiring made from No. 24 hookup wire. Stick-on cable clamps keep the wires in place. A piece of two-wire (No. 26) ribbon cable runs from the +15V power supply to the power-on indicator LED.

The two VM boards are mounted on thin aluminum brackets via ¼-inch spacers. The brackets are slightly larger than the boards, with ½-inch-wide, right-angle flanges bolted to the top panel. I mounted the boards so that all calibration pots

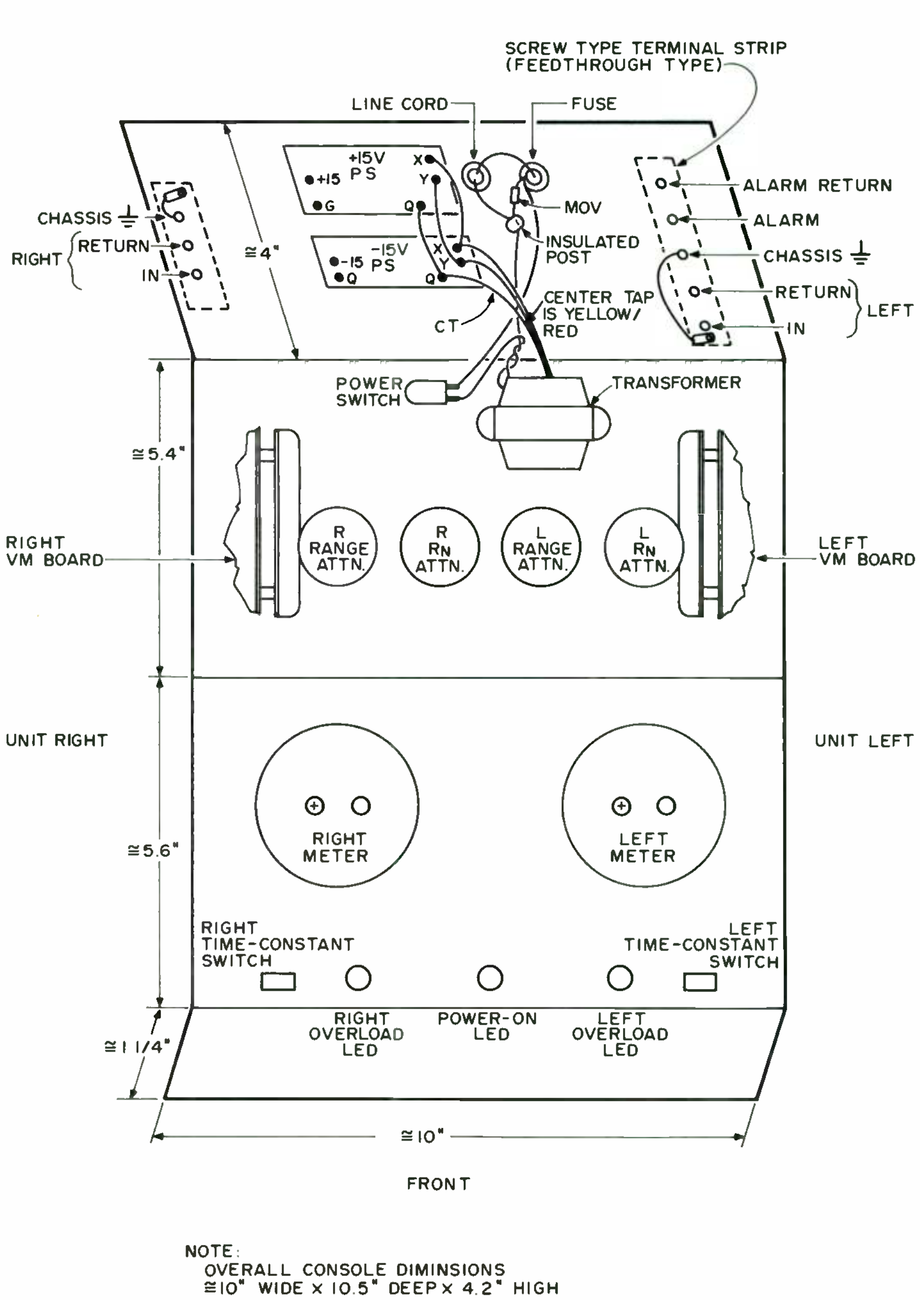


FIGURE II-7: In this diagram of the parts placement for the wattmeter, the top panel is in an open, or flat, position. The "right" designations refer to the unit's right and thus appear on your left as you look at the drawing.

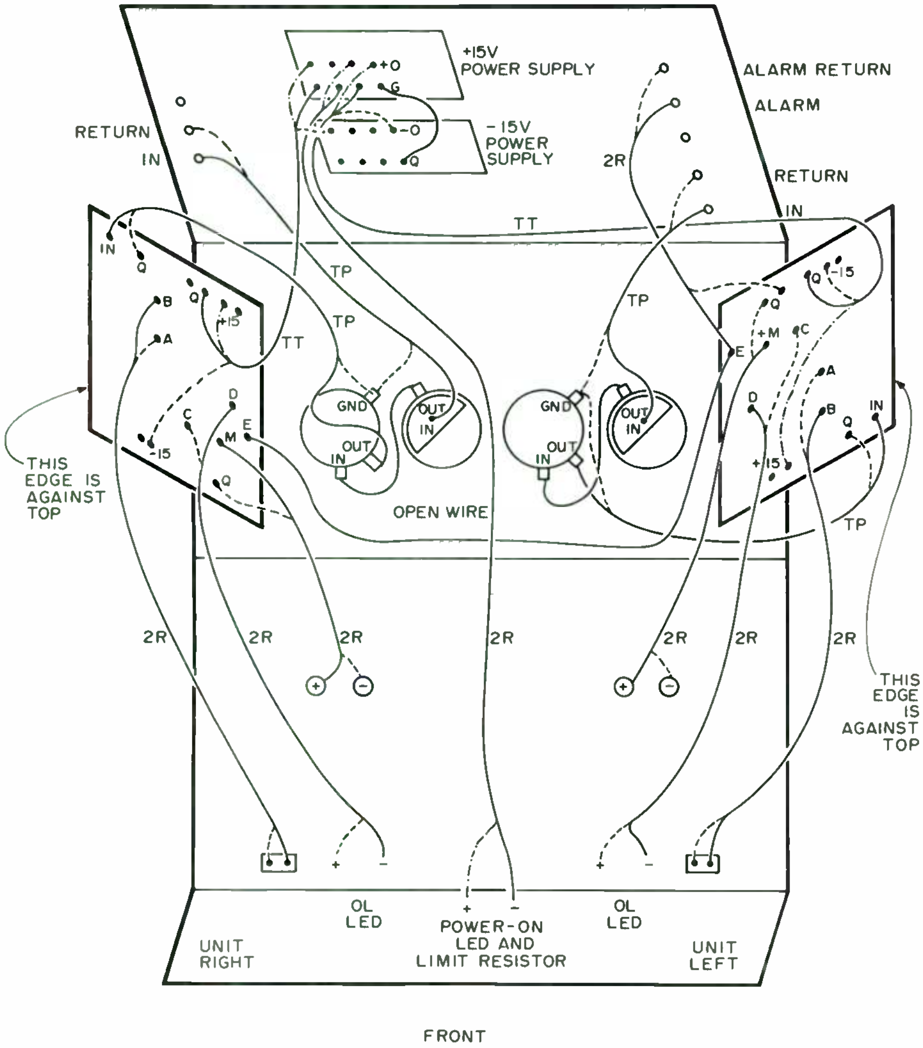
face the bottom of the unit. Individual two-wire ribbon cable connects them to the meters, time-constant switches and overload LEDs. (See *Fig. II-8.*) In this drawing I have tipped the two VM boards (component side up) inward and pushed them to the outside to show connection points. Exact overload and power-on LED connections are not shown, but the following discussion gives more detail about them and the limiting resistor for the power-on LED. Most miniature toggle switches do an internal mechanical inversion, so the switch should be "closed" when the handle points to the "slow" side.

My LED mounting is slightly different from the norm. I make my own mounts, which are actually circuit boards. They cost me close to nothing because I make them as fillers for unused portions of board layouts. If you decide to go this route, *Fig. II-9* shows a negative of four mounts. *Figure II-10a* shows how they are mounted to a panel. These mounts have a provision for mounting two resistors in series with the LED. I also included a limit resistor on the green power-on LED mounting board. (See *Fig. II-10b.*)

My construction of the R_N attenuator is also a little different. This attenuator uses some power resistors that have one end (the input) in common. (See *Fig. II-3.*) I cut a scrap piece of circuit board into a semicircle (approximately 1.3 inches in diameter) and drilled holes to match the resistor leads (*Fig. II-6*). The resistors that comprise the R_N attenuator mount to the switch terminals at one end and to this circuit board at the other.

I constructed the range attenuator by mounting the resistors with "unused" switch terminals. I mounted the input terminal strips vertically at each end of the rear wall and wired them to the R_N attenuator with twisted pair No. 24 wire. Note in *Figs. II-4* and *II-8* that the R_N attenuator does not interrupt the return side of these wires. The R_N attenuator is wired in series with the high side wire. I used the circuit board as input and the switch wiper as output. The output from the R_N attenuator goes to the range attenuator, then (via twisted pair) to the VM input and ground. An open wire ties the alarm (E terminal) outputs together on the two VM boards, and a two-wire ribbon takes the alarm and return to the left-channel terminal strip. Although the positioning and length of the wires are not critical, I would suggest that you use twisted pair wiring from the input terminals to the VM boards and that you keep wiring to the time-constant switches reasonably short.

SERVICING. The best way to make a piece of equipment fail is to build it so that it is unserviceable. To facilitate servicing, you should leave some slack in the wiring from the transformer to the power supply boards and in the DC ouput wiring from these boards. This will allow you to unmount the boards and swing them out together for servicing. The way I mounted the LEDs



KEY:
TP = TWISTED PAIR
TT = TWISTED TRIPLET
2R = 2 WIRE RIBBON

NOTE:
COMPONENT SIDE VIEWS OF VOLTMETER BOARDS

FIGURE II-8: Wiring of the wattmeter.

and time-constant switches allows you to unmount them also. Doing this and removing the meter connections makes it easy for you to unmount the VM circuit boards and swing them out. Leaving some slack in the power, input and alarm wiring to these boards also helps.

The main restriction is the open wire that connects the two alarm outputs on the VM boards, but you can cut and reconnect this wire if you must. Only a few wires connect the four attenuator switches and their resistors. If they need service, unmount them, then disconnect the wires and remove the parts.

In two years, my wattmeter has failed only once, when one overload LED developed an internal mechanical intermittent. I replaced it easily. I have checked calibration twice in

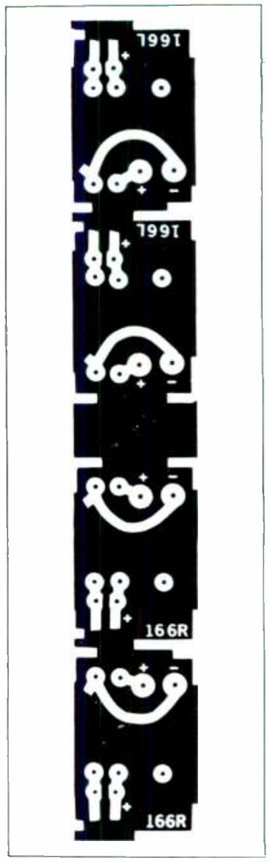
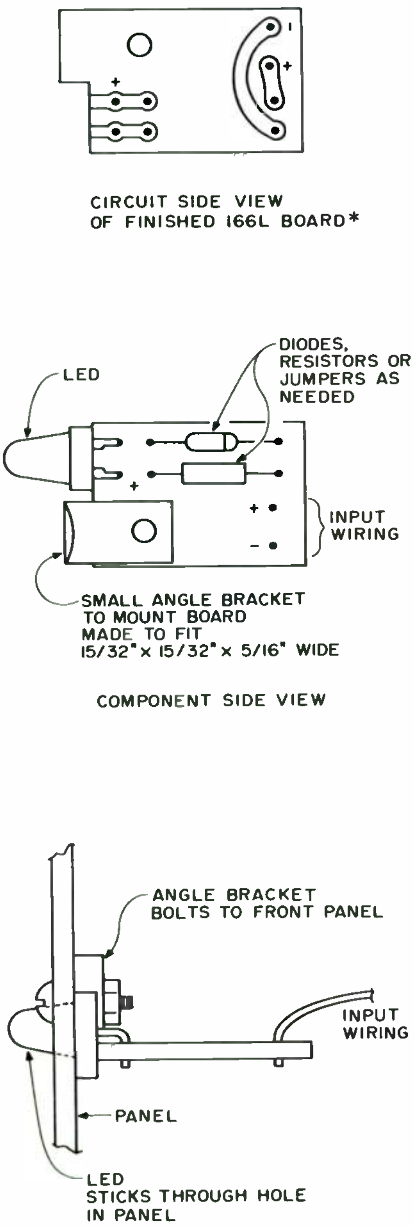


FIGURE II-9: Circuit board for the LED mounts.

that time, but the meter has needed no readjustment.

CALIBRATION. The only part requiring calibration is the VM circuit boards. I discussed this in Part I, but since then have changed the averager gain. To calibrate the finished wattmeter for sinusoidal signals, adjust P3 on each VM board so that the voltage at test point (TP) 2 is approximately 4.5V. This raises the meter protection limit to allow board calibration. Adjust P2 to maximum resistance (full counterclockwise) and set R_N to 4Ω and the range to 1W full scale. Under these conditions, the input terminals are directly connected to the VM input. Apply a sinusoid of 2V RMS at a convenient frequency (around 1kHz). With the time-constant switch set at fast, adjust pot P1 for 4V DC at TP-2. Now adjust P2 until the meter reads full scale. Adjust P3 to drop the voltage at TP-3 to about 4.1V DC. Raise the input slightly to verify that the overload protection is working prop-



*THE 166R BOARD IS A MIRROR IMAGE OF THIS.

Note: Indicated angle brackets have one leg curved. The curved side goes against the panel, and the flat side against the board. The angle brackets are G. C. Electronics (Rockford, IL 61101) catalog number 11-4000-C (old number 6261-C). They are available from Edlie Electronics, 2700 Hempstead Tpke., Levittown, NY 11756.

FIGURE II-10a: Mounting of the LEDs with small circuit boards. Note the provision for mounting two resistors in series with the LED.

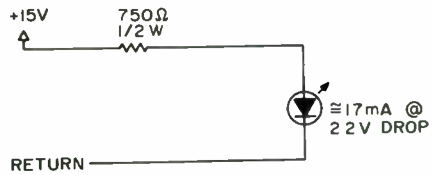


FIGURE II-10b: Of special interest here is the limit resistor on the green power-on LED mounting board. See Fig. II-9 for the board negative.

erly. Doing this on both channels completes calibration.

Use the same procedure to calibrate your wattmeter channels so they read correctly with Gaussian noise signals, except the input sinusoid should be 1.77V RMS (i.e., $2/(1.27)^{1/2}$) when you adjust P1 to produce 4V DC at TP-2. Set P2 and P3 as above. The meter should now read full scale with 2V RMS Gaussian noise input.

PERFORMANCE. I discussed the performance of the VM board in Part I, but, as I mentioned before, I have changed the averager gain. In addition, there is the question of how accurate the input attenuators (R_N and range) and the homemade meter scales are.

Table II-5 shows the measured performance for my two wattmeter channels with a 1kHz sinusoid input. I adjusted the voltage (V_{in}) with various attenuator settings to produce a given meter reading, then recorded this V_{in} value. *Table II-5* compares the measured V_{in} value with the computed V_{in} to produce the indicated meter reading. For all cases where I took the meter to full scale (10), the test results were within 1 percent. This verifies the accuracy of the R_N attenuators and the 1W and 10W positions of the range attenuator. My oscillator would not produce enough output to get to full scale on the 100W range.

This accuracy is better than the design would indicate because of the VM boards' loading on the attenuators. This is a case of cancelling errors, I suspect, so you should not plan on 1 percent accuracy. At 5 on the meter scale, accuracy was somewhat poorer (±2.5 percent), which is probably due to error in making the scale. At 1 on the scale, the accuracy was better than ±1.5 percent, verifying the linearity of the unit.

Figure II-11 shows the frequency response of the right wattmeter channel around full scale (10 on the meter) and at 10dB down (1 on the meter). Note the expanded scales used in these plots. To get data with this high a resolution, I read the voltage at TP-2 on the VM with a digital voltmeter (DVM). I watched the meter to make sure it was tracking this DC voltage. From 10Hz to 20kHz, the response stays within ±0.25dB. Undoubtedly, this unit

TABLE II-5

PERFORMANCE OF COMPLETED WATTMETER

Attenuators		Meter	Computed	Left Channel		Right Channel	
R_N	Range	Reading	V_{IN}	V_{IN}	% Error	V_{IN}	% Error
4Ω	1W	10 (1W)	2	1.99	−0.5	2.00	0
4Ω	1W	5 (½W)	1.414	1.38	−2.4	1.395	−1.34
6Ω	1W	10 (1W)	2.45	2.45	0	2.46	+0.41
8Ω	1W	10 (1W)	2.83	2.84	+0.35	2.84	+0.35
12Ω	1W	10 (1W)	3.46	3.49	+0.87	3.48	+0.58
16Ω	1W	10 (1W)	4	4.04	+1	4.04	+1
4Ω	10W	10 (10W)	6.32	6.34	+0.32	6.38	+0.95
4Ω	10W	5 (5W)	4.47	4.39	−1.79	4.44	−0.67
4Ω	100W	1 (10W)	6.32	6.33	−0.16	6.40	+1.27

Tests at 1kHz with sinusoidal input. All voltages are RMS.

$$\% \text{ Error} = \frac{\text{Actual } V_{IN} - \text{Computed } V_{IN}}{\text{Computed } V_{IN}} \times 100\%$$

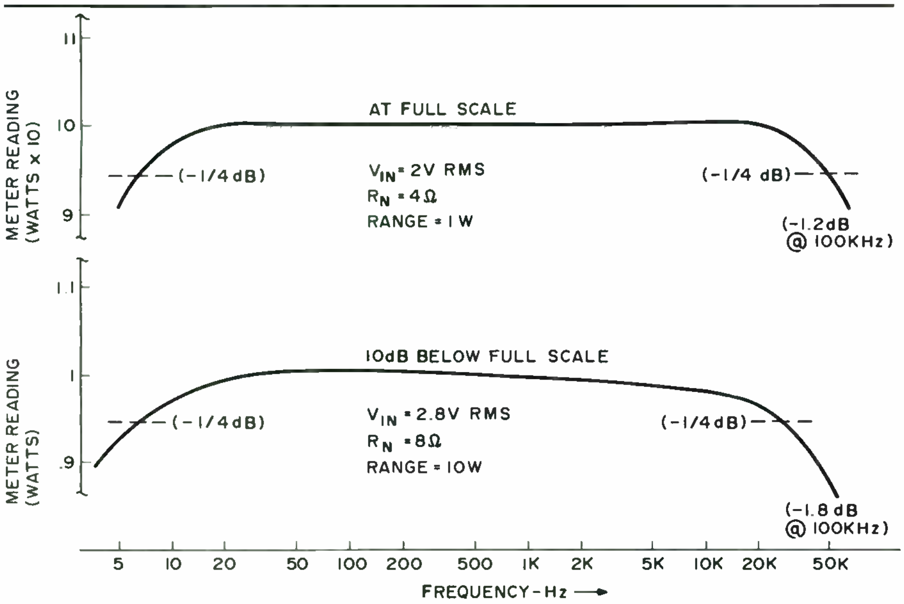


FIGURE II-11: Frequency response of the right wattmeter channel in the fast mode set at around full scale (10 on the meter) and at 10dB down (1 on the meter).

will show interference or oscillation problems up to 100kHz, as the response is down less than 2dB at 100kHz. After a sudden 10dB signal input change, the wattmeter took two or three seconds to settle in the fast mode and 14 to 16 seconds to settle in the slow mode.

The modification to the VM averager gain in the wattmeter will improve the full-scale performance with high-crest factor (peak-to-RMS ratio) signals. I don't have the necessary test equipment to provide actual data, but the first stage of the VM should be capable of at least a 12V peak swing, so the wattmeter should have good linearity with signals having crest factors up to about 16dB. A 14dB crest-factor capability is usually considered sufficient for noise signals.

OPERATION. The dual-channel wattmeter is connected to the outputs of your test amplifier(s). I usually use the two amplifier channels in the normal left-right stereo mode. In the slow mode, the wattmeter gives you an idea of how much power you are putting into each speaker with program material. If you set R_N at or below your speaker's minimum impedance, the true average power delivered to the speakers will be less than the power indicated. Remember that peak power will be much higher.

To run steady-state acoustic tests with CW, noise or warble tone inputs, you would set R_N to the nominal impedance of the system and use the fast mode to allow easy signal-level adjustment. Remember, at certain frequencies you might actually deliver more watts to the system than is indicated on the meters, but it is unlikely that your ears will allow excessive power inputs during steady-state testing.

I sometimes convert my two amplifier channels to a monaural biamp setup with an electronic crossover. I can then monitor the power to the drivers on each side of the crossover frequency. It is interesting to see how the power divides with various types of music. This information is useful in helping to select which crossover frequency to use if the drivers will allow it to move or if one driver has a low power rating.

In Part III of this series, the author will discuss the development of circuit boards that measure phase angle.

REFERENCES

1. J.R. Hunt, "Audio Power Meter," *Popular Electronics*, October 1979, p. 62.

A strongly opinionated cautionary tale of one man's fifteen-year search for the ultimate home built speaker system

A SPEAKER BUILDER'S ODYSSEY: PART V

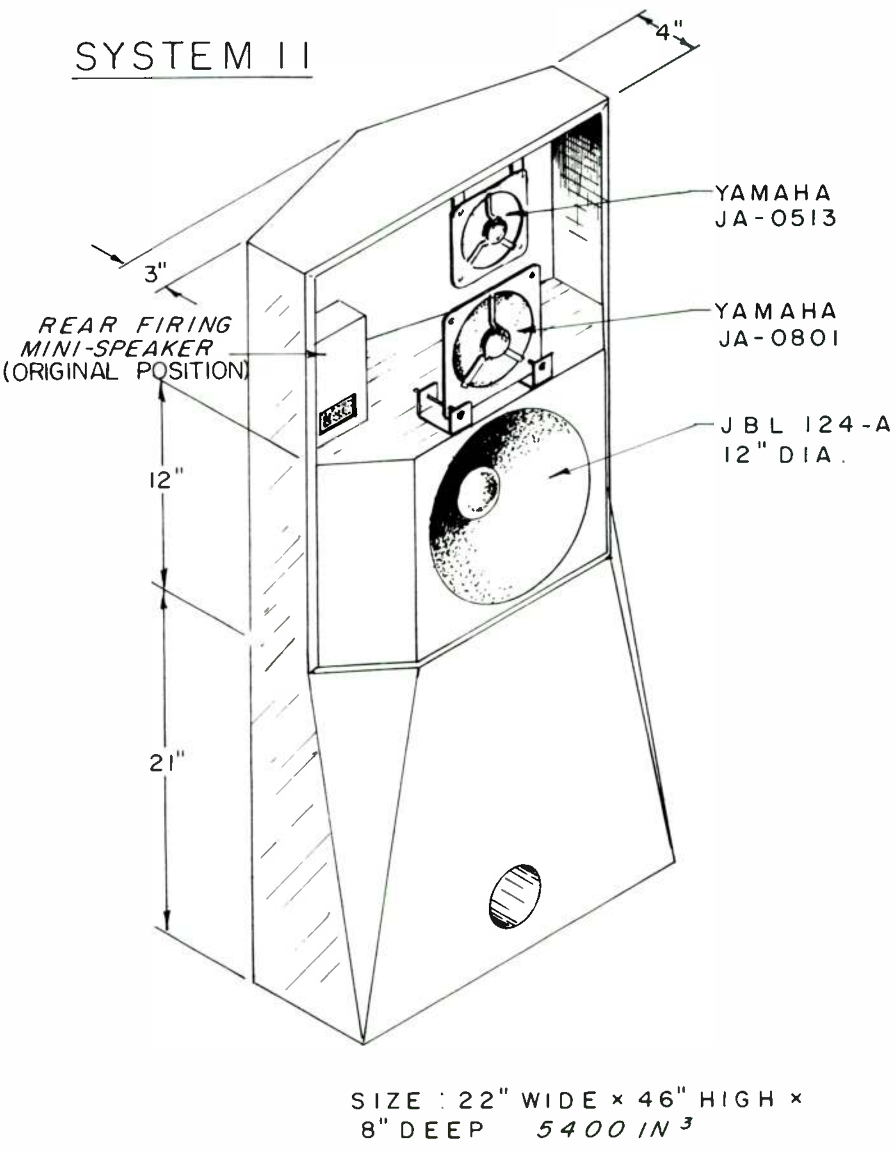
BY ROBERT CARLBERG

SYSTEM 11: MARK ELEVENS. Having learned my major lessons already, I believed all that remained was to achieve that last bit of quality difference in driver and visual design. I set out, theory in hand, to construct the ultimate home-built speaker system. The alternative would have been to give up speaker building and buy a megabuck electrostatic system, which I could not adapt and certainly could not afford.

I already had the "ultimate" woofers on hand, so I went shopping for midranges and tweeters. Thinking back over all the systems I had heard and all the drivers I had tested, only a few possibilities stood out. The Dahlquists were good, but consisted of four different drivers of varying availability requiring an elaborate crossover network. It was difficult to find out exactly what was inside the Snells, but they turned out to be unavailable anyway.

An electrostatic high-end would have been nice, but the cost was high and auditions of separate component systems not readily available. The ESS Heil driver was terribly expensive and not guaranteed to be free of the colorations encountered in ESS's applications. Even more exotic designs such as glass plates and plasma drivers were much too costly to buy unheard.

The only other possibility was a high-tech version of an old idea, the dome. Yamaha's beryllium-dome mids and tweeters from the NS-1000 system were impressive, even



shackled to a second-rate woofer in a fifth-rate cabinet design. On investigation, I found my local Yamaha dealer (stereo, not motorcycle) was willing to special-order them as replacements. They are not, strictly speaking, on the separate component market. At $195 per side, they weren't cheap, but they weren't beyond possibility either. The way I justified it to myself was that this was still considerably cheaper than buying a comparable commercial design.

You may have noticed I prefer dome drivers. This is for two reasons—dispersion and freedom from break-up. Cone drivers will "beam" at their upper frequency limits, and the passive cone will begin to lag behind the active voice coil, causing cone "break-up" at high frequency or amplitude levels. Domes are, I believe, less susceptible to these problems, at least they sound that way.

I further believe that horn drivers are undesirable for the same reasons that "electrical" microphone recording replaced the older "acoustic" recording, which used horns to capture the sound. (Read the back of a computer-enhanced Caruso record.) Horns provide an inherent coloration by emphasizing certain frequencies and de-emphasizing others, but with an overall gain in efficiency. I've never heard a horn that didn't sound like "a horn." Finally, horn drivers cannot be time-aligned due to their convoluted sound paths. Believe me—I've tried.

A visual analogy might clarify this. The driver is a moving surface attached to a voice coil in a magnetic field. The voice coil translates alternating current into alternating sound pressure—an exact reverse analog of the recording process with a microphone. Ideally, the long electro/mechanical chain of events between the microphone and speaker diaphragms should disappear into zero distortion. What is left? An enlarged microphone diaphragm in your living room, re-creating recording studio motions. Why would you encumber it with a horn, point it at the floor, wall or ceiling, or electronically alter the signal going into it?

So while awaiting my specially ordered Yamaha drivers, I began cabinet construction. I altered the Full-Range Panel enclosures to experiment with JBL woofer loading. At 1 cubic foot they sounded restrained, at 2 less so, at 3 even less, but at 4 they were not appreciably improved over 3 cubic feet. The woofers are so closely controlled that foam stuffing affects them very little, except directly behind the woofer for standing-wave cancellation. I therefore settled on 3-cubic-foot versions of the basic design I had built for Marc, with foam only as needed inside.

Since this was to be "the ultimate" cabinet, I decided to find something more rigid than ¾-inch particle board. After several weeks of searching, I finally located two 4×8's (one already cut into two 2×8's) of 1-inch particle board at a local lumber yard. The clerk was glad to sell this special order someone had never picked up. But did you ever try lifting and carrying 32 square feet of 1-inch particle board?

The cabinets went together with a minimum of problems, aided by a borrowed table saw, liberal rasping and Weldwood plastic resin glue to fill the gaps. The overall dimensions were held to within about 1/32 inch of design (mostly for my own amusement), but each cabinet had a wide variation in how these dimensions were met. They were about two months in the making, with my working evenings and weekends, after which I spent another four weeks veneering the cabinets with real walnut.

I no longer had access to laminate at employee discount (which put it out of my price range), and since this was to be a visual adventure, I decided to try the next logical step. It went on similarly to the laminate, except I found that veneer shrinks slightly when subjected to the moisture in the glue, leaving me with hairline cracks between the larger expanses. Oh well, from a distance it looks good.

In addition to being designed for high strength through the use of small surfaces wherever possible, two internal braces run the length of the cabinets, bisecting the rear panels and trisecting the front. The rear panels are angled to lessen standing waves. After cabinet depth, the primary objective was stability—and carrying them upstairs convinced me they would remain stable thereafter.

Next time, Mr. Carlberg will continue this discussion of his Yamaha speakers.

Tools, Tips & Techniques

Improved Imaging

Most serious stereo listening is done at positions equidistant from the two speakers. This defines the optimum listening area. Even small lateral shifts in position will produce shifts in the stereo image. The following is a simple, effective and (best of all) cost-free way of broadening the optimum listening area using conventional speakers.

In *Fig. 1*, position 1, the speakers' contributions are equal. Center stage voices are localized precisely between the two speakers. In position 2, however, center stage voices are shifted to the right. The left speaker's direct energy is attenuated because the listener is farther away from it and farther off axis. The direct energy is also delayed relative to that coming from the right speaker because of the difference in path lengths. As the listener moves from position 1 to 3, the center stage follows, and the stereo breadth shrinks until all the sound appears to be coming from the right speaker.

You can compensate for this effect by angling the speakers sharply inward,* as shown in *Fig. 2*. At position 3, the direct energy from the left speaker still arrives late, but its amplitude over a wide band of frequencies is at least equal to that coming from the right. This is because you are listening nearly on axis relative to the left speaker, while you are well off axis relative to the right.

I made my measurements in an average room using a pair of 10-inch, two-way speakers. I set them up as in *Fig. 1*, with a microphone at position 2, and recorded the contributions of each speaker as a function of frequency. Note that the contribution of the right speaker is considerably greater than that of the left (*Fig. 3*).

Next, I angled the speakers, as in *Fig. 2*, with θ equal to 50°. Note that the con-

Continued on page 43

*This concept was introduced by Benjamin B. Bauer in a patent granted March 5, 1963, and assigned to CBS. I am not aware of any loudspeaker that uses any embodiment of the patent or any concept the patent may have introduced during its 17-year duration. Recently, however, several companies have instituted variations of Bauer's basic concept, most notably Bose in its new Delco/GM-Bose automotive systems.

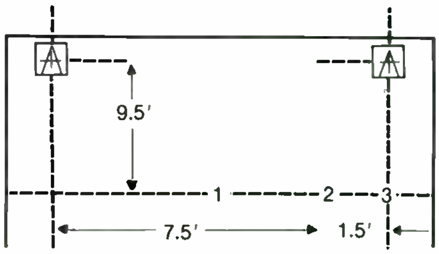


FIGURE 1: When you position your speakers straight-on, the center stage follows you as you move from position 1 to 3, and the stereo breadth shrinks until all the sound seems to come from the right speaker.

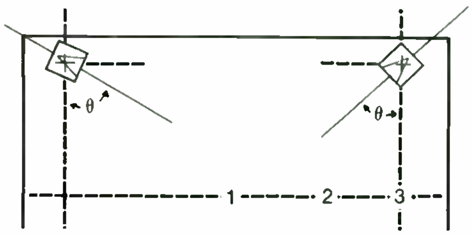


FIGURE 2: To compensate for the shrinking that occurs in Fig. 1, angle the speakers sharply inward.

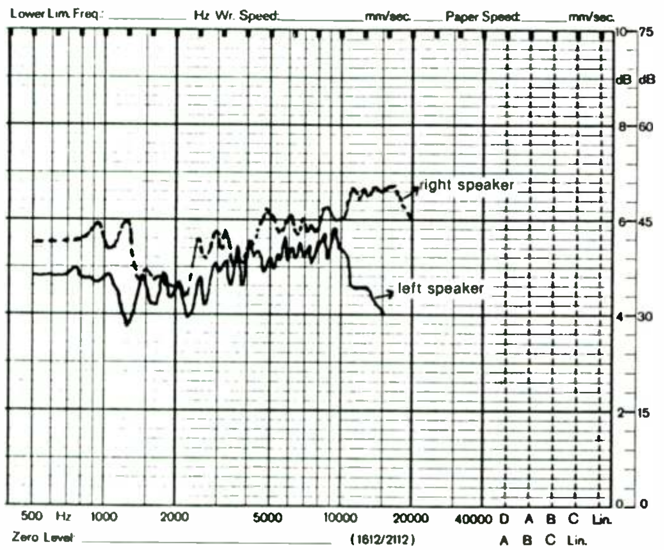


FIGURE 3: Frequency readings of the speakers in the standard, straight-on arrangement at position 2.

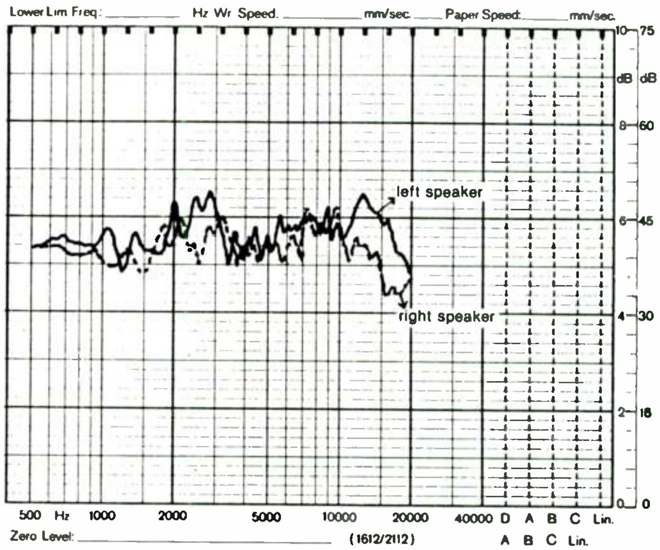


FIGURE 4: Frequency readings of the speakers in the angled arrangement at position 2.

Craftsman's Corner

Designer Speakers

These Harry F. Olson-inspired speakers are the work of Canadian cabinetmaker John Otvos, who spent roughly 250 hours handcrafting the pair. The solid red oak cabinets are made of 15/16-inch timber and are internally damped with plastic tar and ½-inch "tootie-frootie" carpet underlay. Walnut feet, 160 feet of brass inlay, and a rosewood and satinwood conch inlay add to the elegance of the cabinets.

Inside, each speaker's driver complement consists of a 14-inch Audax woofer, with its magnet braced to the cabinet, a 5-inch Audax midrange with a separate spherical cabinet, and two 1-inch Seas ferro-fluid tweeters that are wired in parallel. All magnets and baskets are damped with flexible refrigeration compound. Otvos used a Philips crossover and Oracle wiring (720 strand) throughout.

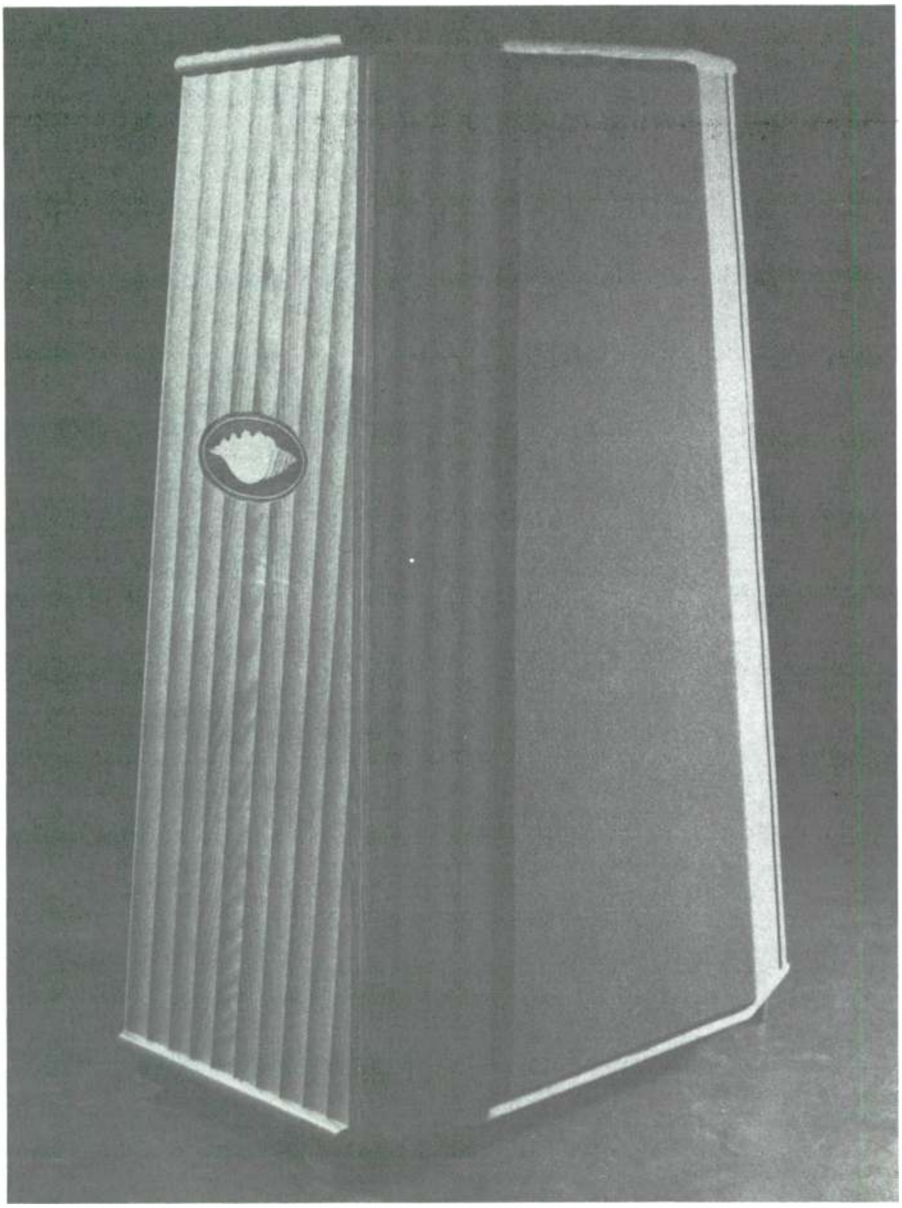


PHOTO 1: Otvos's solid red oak speakers, which he describes as "extremely open and natural sounding," were originally housed in a 20-by-30-foot (600-square-foot) room with a 19-foot cathedral ceiling. "It was my personal concert hall," he writes.

PHOTO 2: Otvos recently added black felt to the driver baffle, which is made of ¾-inch particle board and reinforced with heavy-duty plastic laminate.

The speaker's voice coils are aligned and offset ½ inch. For better sound reproduction, the left speaker is a mirror image of the right, and both are designed with nonparallel sides, front and top. The driver baffle is made of ¾-inch particle board, which is reinforced with heavy-duty plastic laminate and covered with black felt. Four-inch-thick fiberglass batting is mounted on the back wall, while a 3-inch triangular foam baffle is mounted off-center behind the tweeters and suspended from the speaker top. The brass screws that attach the speaker back to the sides, top and bottom of the cabinet are all polished and covered with clear lacquer. A ¾-inch acoustic foam grille cloth, supported along the perimeter with Velcro, adds the finishing touch.

These exquisite speakers, which are 52 inches high, 24 inches deep, 32 inches wide at the base and 26 inches wide at the top, have an internal volume of 6.75 cubic feet. Otvos estimates their value at $5,000 for the pair.

John Otvos
Brighton, Ontario
Canada K0K 1H0

SB Mailbox

UPDATE: ACOUSTICAL PHYSICS

*Bill Morrison of Acoustical Physics has pointed out a couple of misleading items in our review of the Acoustic Image Model II loudspeaker (*SB *3/83, p. 34). The bass inductor in* Fig. 1 *is not iron but ferrite cored, which is powdered iron glued into a rod shape. According to Bill, the new resistor to balance the midrange should be a 20 ohm value (10W), rather than the 5 ohm one I suggested.—Ed.*

CORRECTION: 3M STILL KICKING

I am writing in reference to a letter from Kenneth M. Rauen (*SB* 2/83, p. 33) stating, "Technidisc is the only American producer of laser video disks—since IBM and MCA merged, and DiscoVision Associates (DVA) and 3M went out of the business...."

I do not know where Mr. Rauen received this astounding information, but I'm sure our hundreds of customers would heartily disagree with him. 3M is not only in the laser video disk business, but now offers one and three-day turnaround on limited production runs.

As a matter of fact, it is interesting to learn Technidisc is finally *in* business. I hope their disk production is more error-free than Mr. Rauen's letter.

Frank M. Price
Sales & Marketing Manager
Optical Recording Project/3M
3M Center
St. Paul, MN 55144

Mr. Rauen replies:

I thank Mr. Price for his letter saying 3M is alive and well. I am sorry I gave SB *readers false information. I received that information from a supposedly authoritative source within Technidisc. I have since left Technidisc and am negotiating with 3M for a process development position. Once again, I apologize for my error.*

REBUTTING BALLARD

I am concerned about Mr. Ballard's article on electronic crossover design (*SB* 3/82, p. 14; 4/82, p. 26). In addition to some minor conceptual errors in the properties of crossover networks, he makes three major errors in his version of the three-way, fourth-order Linkwitz-Riley (LR4) crossover network. These errors nullify the subtle (but real) benefits he seeks to gain from these filters. The three errors are:

(1) Incorrect interconnection of the filters for a three-way LR4 crossover. The correct topology is noninteracting and needs no experimental "tweaking."

(2) Incomplete compensation for inherent phase errors in the three-way LR4 crossover. (This is really part of the first problem.)

(3) Improper use of phase equalization to align the acoustic centers of the individual drivers.

To critique Mr. Ballard's article properly, I must present some background material on "ideal" loudspeakers and crossovers. The discussion will be somewhat long-winded and slightly mathematical, so please bear with me.

The "ideal" loudspeaker must have two properties—flat magnitude response and linear phase response. It should also have an acceptable polar response. Let's define the first two properties a bit more carefully. *Flat magnitude response* means that the acoustic sound pressure level (SPL) at a fixed point in space must be constant, as the frequency of a constant amplitude electrical sine wave input to the loudspeaker is varied over the audio band. *Linear phase response* means that the phase angle between the sine wave electrical input and the sinusoidal acoustic output at a fixed point in space must grow linearly with increasing frequency. The relationship between the electrical input and acoustic output of the ideal loudspeaker (often called the loudspeaker transfer function) is shown in *Fig. 1*. Mathematically, you can express the magnitude characteristic, G, and phase characteristic, ϕ, of the ideal loudspeaker as follows:

$$G(f) = G_o$$

$$\phi(f) = -2\pi T_o f$$

where G_o and T_o are constants and f is frequency in Hertz. (Note that $\phi = o$ is a special case of linear phase.)

Let's now define two types of distortion. *Amplitude distortion* in a loudspeaker is simply its departure from flat response. *Phase distortion* is its departure from linear phase response.

An alternate property of interest is *time delay*. The time delay, T, experienced by a sinusoidal signal passing through a loudspeaker system with phase lag, ϕ, is

$$T(f) = \frac{-\phi(f)}{2\pi f}$$

In general, this time delay varies with frequency. For a system with linear phase, however, the delay is

$$T(f) = \frac{2\pi T_o f}{2\pi f} = T_o = \text{A CONSTANT}$$

Thus, you see that a loudspeaker with linear phase delays all incoming frequencies equally. Such a speaker has no "time smear." If the speaker also has flat frequency response, it has perfect "impulse response" and no "transient errors."

Now, for a multiple-driver loudspeaker to have perfect impulse response, the acoustic output of all the drivers must add together (properly considering amplitude and phase) to yield the flat amplitude and linear phase response of *Fig. 1*. Assuming for the moment that the individual drivers in the multiway speaker are "ideal" over their respective frequency ranges and that their acoustic centers are physically aligned, then the electrical responses of the ideal crossover networks must also add electrically to produce flat frequency response and linear phase response.

One class of crossover networks that has this response is the "sum-to-one," or "constant voltage," crossover. Small[1] discusses several forms for these crossovers. None of these filters has the high-pass and low-pass outputs in phase. This leads to the first conceptual error in Mr. Ballard's article: the absence of phase distortion does not imply or require that the individual drivers be in phase at the crossover frequency (or anywhere else).

Constant voltage crossovers are not suitable for all multiway systems, since the ultimate attenuation rate in at least one channel is limited to 6dB/octave. Recognizing that we are more sensitive to amplitude response distortion than phase distortion, a compromise set of crossover networks has evolved. When summed together, these constant magnitude response filters provide flat frequency response and higher roll-off rates, but do not have linear phase. Included in this class of crossover are the following:

(a) all odd-order Butterworth filters

(b) all cascades of two Butterworth filters.

Category (a) crossovers have these properties:

- response down 3dB at cross frequency (f_c)
- outputs 90° out of phase at all frequencies
- constant output power
- roll-off rates of 6, 18, 30, etc. dB/octave

Category (b) crossovers have these properties:

- response down 6dB at f_c
- outputs in phase at all frequencies
- one-half total power at f_c
- roll-off rates of 12, 24, 36, etc. dB/octave.

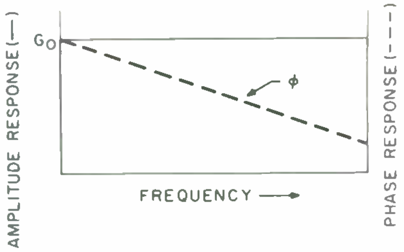


FIGURE 1: In an "ideal" multiple-driver loudspeaker, the acoustic output of all the drivers must combine to produce flat amplitude and linear phase response.

The in-phase property of category (b) filters is desirable because it eliminates frequency-dependent tilt in the polar radiation pattern of noncoincident drivers. (See Linkwitz, *SB* 3/80, p. 9.)

Any network that has flat frequency response with a frequency-dependent phase shift is an all-pass network. Thus, the constant magnitude crossovers sum to the all-pass network response. None of these crossovers has linear phase, and thus all of them produce some frequency-dependent time delay, or "time smear."

Mr. Ballard's fourth-order filter was popularized by Linkwitz[2] and *SB*. The outputs of a two-way Linkwitz filter sum to a second-order all-pass network denoted ϕ (f_o, Q) with center frequency f_o equal to f_c and a Q of 0.707. (Ballard's *Fig. 5* shows an op amp realization of a second-order all-pass.) The time delay of a two-way, fourth-order Linkwitz filter and a third-order Butterworth filter are shown in my *Fig. 2*. Note that the Linkwitz filter produces a sharp change in time delay through the crossover region relative to the third-order filter. (Linkwitz never said his crossover was free of phase distortion.) Some evidence[3] suggests that this time smear is discernible in some types of program material when crossing over in the 100Hz to 3kHz region.

This relates to Mr. Ballard's first two

major errors. (In Ballard's defense, Linkwitz makes the same errors.) You must cascade the constant-amplitude response filters for multiway systems in a specific manner to maintain the overall flat frequency response property. When they are correctly interconnected, no experimental "tweaking" of individual filter responses "a la Ballard" is required. Furthermore, when they are improperly connected, no simple *ad hoc* "tweaking" will bring them back to flat-summed response.

Without going into tedious mathematical detail, the correct three-way topologies are shown in *Figs. 3a, b* and *c* using Ballard's crossover frequencies as an example. The three-way summed response, shown in *Fig. 3d*, is a cascade of two all-pass networks. Ballard's topology appears in *Fig. 3e* for comparison.

In *Fig. 3a*, notice that E_{MID} and E_{LOW} are in phase and 6dB down relative to E_1 (not E_{IN}), so they are in the proper phase and amplitude relationship for the 400Hz crossover. E_2 has the correct magnitude relationship relative to E_{MID} and E_{LOW}, but you must correct it for the additional all-pass phase response suffered by $E_{MID}+E_{LOW}$. You can build *Figs. 3a* and *3b* with nine op amps, two for each filter and one for the all-pass. *Figure 3c* makes direct use of the sum-to-all-pass property to synthesize the network by subtraction. You can build this form with eight op amps.

You should be aware of some practical considerations. First, if the crossover frequencies are widely separated (i.e., a factor of ten or more), you can probably eliminate the all-pass phase correction circuit with no audible effect. Under the same condition the Ballard/Linkwitz topology will also perform satisfactorily, but as I said before, it makes no sense to "tweak" the filters. Second, any additional frequency shaping (e.g., rumble and hiss filters) must occur before E_{IN} to preserve the constant magnitude property of the crossover. Finally, if the crossover frequencies are close, the individual outputs might not be exactly 6dB down and in phase because all three drivers are contributing to the output. *Figures 3a, b* and *c*, however, will always have flat-summed frequency response without experimental "tweaking."

My final comment concerns the proper use of phase equalizers to correct for offset in the acoustic centers of multiple drivers. The purpose of these equalizers is not just to put the drivers in phase at the crossover frequency, but to provide a constant time delay, equal to the inter-driver's offset, throughout the active crossover region. This preserves the flat frequency response property of the summed outputs over the crossover region. With the proper time delay, the drivers are in phase at all frequencies. Putting them in phase at f_c, however, does not guarantee the correct time delay. The difference, mathematically, is that time delay is unique, while phase is not.

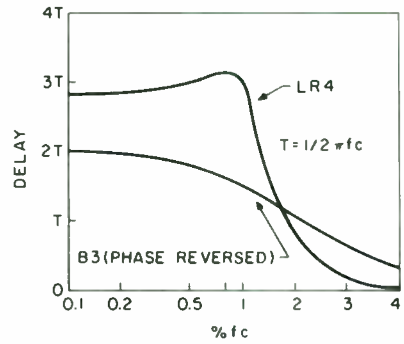


FIGURE 2: The Linkwitz two-way, fourth-order filter (LR4) produces a sharp change in time delay through the crossover region relative to Butterworth's third-order filter (B3).

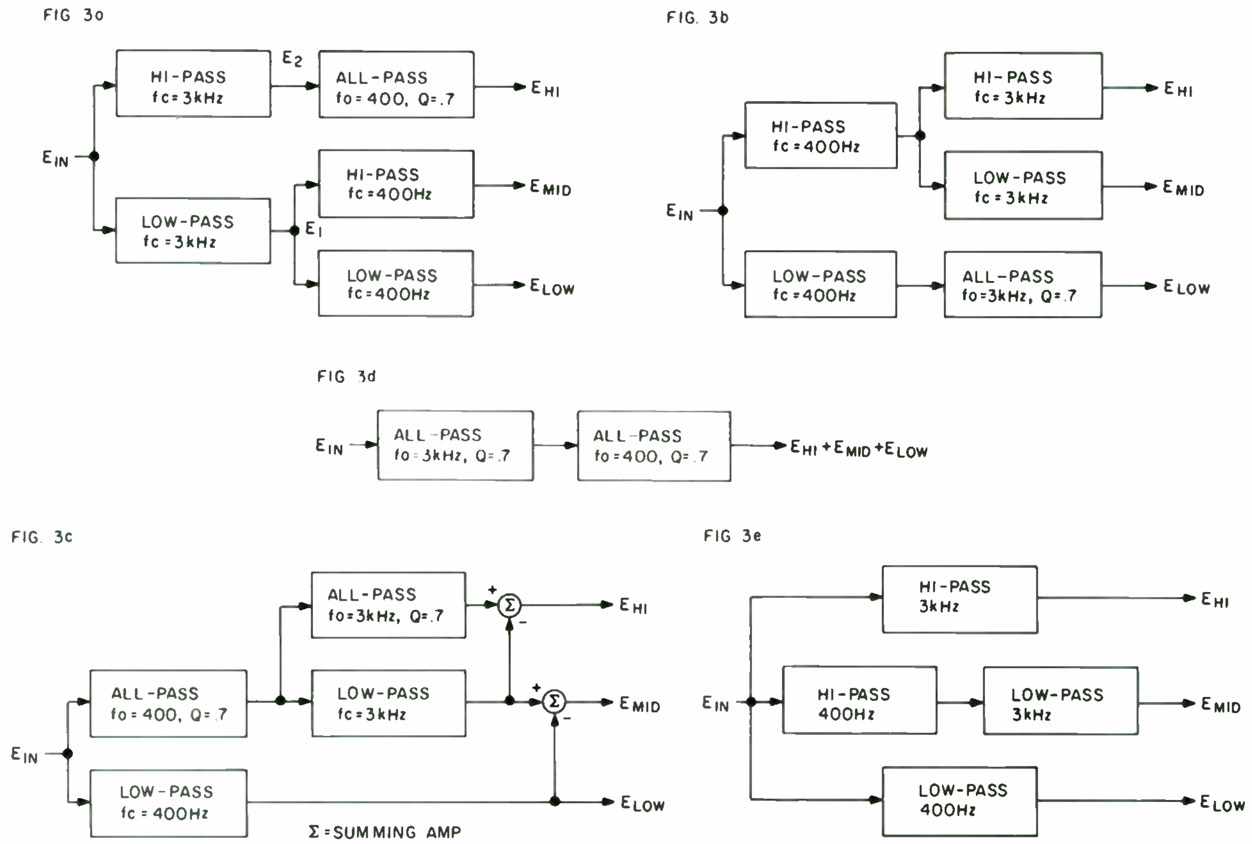


FIGURE 3: Figures 3a, b and c show the correct three-way topologies using Ballard's crossover frequencies. Figure 3d shows the three-way summed all-pass equivalent. Compare this to the Ballard/Linkwitz topology in Fig. 3e.

If you time align two drivers initially, their outputs are in phase at all frequencies. If you have one driver forward or backward a distance equal to one wavelength at 3kHz, the drivers will still be in phase at 3kHz when measured as Ballard recommends (because phase repeats itself every wavelength in distance), but they will be out of phase at all other frequencies (except at 6, 9, 12, etc. kHz) and no longer time aligned.

The distinction between phase lag and time delay relates to another major error in the Ballard article. When time aligning drivers, you cannot arbitrarily select the midrange driver as "home base." Instead, choose the driver with the greatest delay as the reference driver and delay all other drivers to match it. This is because none of the electronic circuits can produce negative delays—i.e., you cannot predict the output of a driver before its electrical input arrives.

In a typical three-way system with all drivers mounted on a common baffle, direct-path wavefronts from the tweeter will generally arrive at an on-axis position first, followed by those from the midrange and the woofer. In general, therefore, you must delay all upper-range drivers relative to the woofer. Pulse-testing methods are the best determinants of these delays. If you use a sine wave phase angle approach, convert the measured phase angle to an equivalent time delay at f_c using

$$T = \frac{-\phi_c}{2\pi f_c}$$

Realize the phase angle with a cascade of delay equalizers to minimize nonlinear phase distortion.

An example here will help. A typical tweeter/midrange offset is 40mm. This corresponds to a time delay of 115μsec, which at 3kHz corresponds to a phase shift of 124°. A second-order phase equalizer with a Q of $1/\sqrt{3}$ is particularly attractive, since it has a time delay that is constant within 1% up to 0.5 f_o. The delay for this equalizer appears in *Fig. 4*. A single equalizer will produce the required 124° of phase shift at 0.79 f_o (f_o = 3.8kHz). Notice, however, that the frequencies immediately above this point suffer the greatest time smear. Using one equalizer would destroy the flat-summed frequency response. To obtain a constant delay for one octave on either side of the crossover region, the equalizer center frequency must be at least 4 f_c—i.e., at least 12kHz for our example. Three equalizers, each providing 39μsec delay (f_o = 14.4kHz), will do the job.

Thus, phase considerations alone would lead you to use one equalizer, whereas delay considerations show that you need three equalizers to provide the required delay without excessive phase distortion.

The situation is more complicated with the woofer/midrange crossover, since an equalizer that provides the proper delay at 400Hz using the 4 f_c criteria will introduce nonlinear phase response in the 3kHz mid/tweeter crossover region. Perhaps the best way to obtain this delay is to physically offset the drivers.

My purpose in writing this letter is not to criticize Mr. Ballard, but to point out that current crossover design philosophy and its theoretical underpinnings have reached a high level of sophistication. Incomplete understanding of this theory can lead to incorrect circuits that *appear* correct on the surface. Mr. Ballard's errors are subtle, and their audibility might be marginal, but audio has reached a level of development where subtleties are of primary concern.

Joseph A. D'Appolito
Andover, MA 01810

Mr. Ballard replies:

Mr. D'Appolito seems to be very knowledgeable in his assertions, particularly those concerning "time delay" versus "phase delay" and the proper topology for interconnecting low and high-pass filters to obtain crossovers with constant voltage output. He makes several statements, however, that lead me to assume that he has never applied any of his knowledge.

First, he refers to some "minor conceptual errors" in my article. He might have more than one in mind, since he uses the plural, but he mentions only one: ". . . the absence of phase distortion does not imply or require that the individual drivers be in phase. . . ." This is true, but there must be adequate in-phase voltage vectors to obtain constant voltage over the crossover region. My circuit accomplishes this and also puts the voltage vectors from adjacent crossovers in phase.

Second, Mr. D'Appolito says that I made "three major errors" in my article. According to my dictionary, the phrase "major error" means that the crossover does not work. In his concluding paragraph, however, he says, "Mr. Ballard's errors are subtle, and their audibility might be marginal . . . ." Subtle? Marginal? Might? Mr. D'Appolito apparently does not know what the network sounds like. I know how my version sounds. I've built it. But Mr. D'Appolito never provides any quantitative information about audible effects—of either his network or mine.

Now for the three major errors.

(1) Incorrect interconnection of the filters for a three-way, fourth-order crossover.

If Linkwitz's LR4 crossover is also wrong, why did Mr. D'Appolito wait so long to disclose this? Nevertheless, Mr. D'Appolito's topology, as presented in Fig. 3, *is interesting and deserves some attention on my part in converting it to usable hardware.*

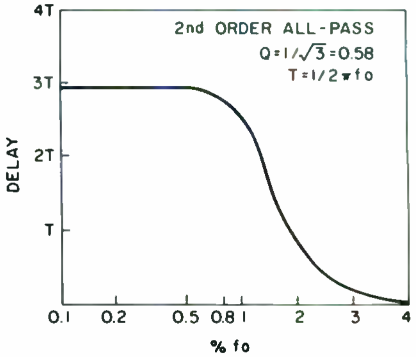


FIGURE 4: A second-order phase equalizer with a Q of $1/\sqrt{3}$ has a time delay that is constant within 1% up to 0.5 f_o.

First, look at Fig. 2, *which shows time delay for the LR4 as practically constant up to f_c = 1 and thereafter encounters a sharp change. Now look at* Fig. 4, *which Mr. D'Appolito implies is the proper way to obtain time delay. It looks very much like the LR4 in* Fig. 2.

In answer to his statement about the practical considerations in his topology, I would say that the same practical considerations apply to my topology. For example, in the bandpass for the midrange, if the two crossover frequencies are four octaves or more more apart (he uses a decade), my circuit also requires no tweaking. I chose 400 and 3,500Hz because they are less than four octaves apart (also slightly less than a decade, but four octaves is a better guideline). Similarly, if I chose 200 and 3,200Hz, which are exactly four octaves apart, the circuit would require no tweaking.

The topology in my circuit, even with tweaking in the bandpass, provides almost flat voltage response, as is proven with a summing amplifier. Bob Bullock, a contributing editor of SB, *has confirmed this through mathematical models with computer printouts, which show deviations of less than 1dB from flat response. Are these bumps of less than 1dB the consequence of a major error?*

(2) My second major error as stated by Mr. D'Appolito is part of the first and thus has been answered.

(3) Improper use of phase equalization.

In his example, Mr. D'Appolito uses a tweeter and midrange with a 40mm offset. How does he measure this? Coil-to-coil centerline? Physical offset does not provide a true picture of phase or time delay offset. To produce an audible output, you must know the acoustical phase centers between drivers when AC voltage is applied. I am sure most audiophiles are aware that a *driver coil moving in its own magnetic field produces a counter emf (a voltage that opposes the applied voltage). Therefore, the complex impedance of the driver coil—represented by inductance and resistance (and sometimes capacitance)—is different when the coil is moving (dynamic) and when it is standing still (static).*

Consequently, sound output from a driver is seldom in phase with applied voltage. Current determines acoustical phase output, but current is seldom in phase with voltage. In addition, physical offset and the driver coil's dynamic impedance affect acoustical phase. Since the combination of these factors is difficult to determine from mathematical analysis, the major feature of my article was to determine actual acoustical phase differences between drivers at crossover frequency and to correct them electrically.

In spite of these "major errors," which might be subtle and might be marginal and possibly can't be heard, I am sure most readers will find vast improvement in their systems with my network. As Mr. D'Appolito implies, however, major improvements are built on an accumulation of subtle improvements, and only actual application of his suggestions will determine whether they produce any audible difference.

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1. Small, R.H., "Constant-Voltage Crossover Network Design," *JAES*, Vol. 19, No. 1, January 1971.
2. Linkwitz, S.H., "Active Crossover Networks for Noncoincident Drivers," *JAES*, Vol. 24, No. 1, January/February 1976.
3. Lipshitz, Stanley P., et al., "Preliminary Results on the Audibility of Midrange Phase Distortion in Audio Systems," 67th Convention of the Audio Engineering Society, New York, October 31 to November 3, 1980 (Preprint 1714 (D-8)).

CONE CORRECTION

We are writing in response to Steve Williamson's letter in *SB* 2/83 (p. 35). A successful method for treating paper cones plagued with clicking dust caps caused by long excursions is to spray the dust caps with Krylon clear acrylic spray paint. Apply the paint in successive layers, allowing for adequate drying between coats, until you achieve the desired stiffness. The acrylic soaks into the fiber and might reduce the cone's susceptibility to environmental changes, particularly humidity.

Ken Rauen
Detroit, MI 48221
Mark Schlorff
Ferndale, MI 48220

MATH SKILLS A MUST

Another round of applause goes to *SB* for G. R. Koonce's recent editorial comment, "A Case for Custom" (2/83, p. 6). In my opinion, if someone is a *true* hobbyist in any field, he or she should take an interest in a project from its inception to its completion. In speaker building, I have found that a project always begins with a string of design formulas and equations, long before the blade hits the wood. Granted, my high school math skills did need some polish, especially after my graduation from "kit" work to custom design, but after reviewing formula rearrangement and consulting several technical compendiums (including those from Speakerlab), most of it was a breeze.

Alas, math is a necessary evil in speaker building, if only to double check a kit or plan, but educational opportunities to hone these skills are cheap and readily available.

Do keep publishing your informative magazine.

Doug Cabaniss
Sullivan, IN 47882

MATH MERITS CONSIDERATION

I endorse Mr. Koonce's editorial in the 2/83 issue of *Speaker Builder* (p. 6). As I see it, the mathematics used to describe the design of speaker systems clearly defines the limits within which any individual must operate. In my own case, I have been able to eliminate several designs that I *thought* were valid, but upon careful analysis found to be off base. By examining the formulas, I have saved a lot of ill-spent money and eventual disappointment.

I agree that the "math stuff" can be difficult and boring, but I have found that many of your authors have taken great pains to make the material understandable to those of us who are not engineers. I especially enjoyed the series by Robert Bullock (*SB* 4/80, p. 7; 1/81, p. 12; 2/81, p. 18; 3/81, p. 18; 1/82, p. 20), which I have used as a tutorial to help me understand the Thiele-Small parameters and loudspeaker design.

In addition, the recent article by Ernest Wittenbreder, "An Audio Pulse Generator" (*SB* 4/83, p. 14), is an excellent step toward the objective evaluation of completed speaker projects. I would also like to see articles on building equipment to evaluate completed speaker systems. *(See G. R. Koonce's series on modular test instruments beginning in* SB *3/83.—Ed.)*

Another enjoyable part of *SB* is "Tools, Tips & Techniques." I would like to see this section expanded, as it includes a lot of creative material.

By the way, if any readers are interested in comprehensive information on making circuit boards, I would suggest a book written by Dr. Joel Goldberg of Macomb County Community College in Warren, Michigan. *How to Make Printed Circuit Boards*, which is published by McGraw-Hill in its electro-skills series, costs $6.50. *(It is available from Old Colony Books, No. MH-2.—Ed.)*

You have a fantastic publication. I hope you continue your successful efforts.

Phillip Trosko
Troy, MI 48099

SANDERS' SURPRISES

I found Roger Sanders's ESL/TL update in *SB* 4/82 (p. 39) quite interesting. Although I have little experience in electrostatics, some of his comments surprised me. First, he is now using two transformers, each having a turns ratio of 45:1. I tried the following formula to calculate the amplifier load at 20kHz:

$$Z_L = \frac{1}{2\pi fC(TR^2)}$$

where f = frequency in hertz, C = speaker capacitance in farads, TR = transformers' turns ratio. Thus, Z_L equals

$$\frac{1}{2\pi(20{,}000\text{Hz})(2{,}400\times10^{-12}\text{F})(45+45)^2}$$

which equals 0.41Ω.
If this is correct, isn't this a pretty severe load for almost any amplifier?

Second, Mr. Sanders states, "Insulating the drivers is unnecessary, since the arcing is no worse without it." I believe that although stator insulation is difficult to install, it is very important in determining maximum sound pressure level (SPL).

Finally, Mr. Sanders suggests using Heath shrinkable diaphragms, which do not allow control of the resonant frequency. Don't you think it is important to be able to control this factor, using a stretcher to place it where you want?

Ricardo O. Lazzari
Buenos Aires, Argentina

Mr. Sanders replies:

Let me take this opportunity to clarify a few *points. First, it would appear that the load is too low for most amplifiers when the transformers are operated in series/parallel as I recommended. I might point out that the load still seems too low even when operated with just one transformer. You are looking at 20kHz, however, where there is very little energy. Down in the lower frequencies, where the work must really get done, the impedance is reasonable. In any event, it really doesn't matter what the math says: amplifiers generally seem to tolerate the load just fine, so don't worry about it.*

As for insulation on the ESLs, because the original insulation techniques were not very good, I decided to do away with them entirely. Actually, better insulation would reduce the SPLs, not increase them. Again, from a practical standpoint, the insulation is not necessary, and the highest SPLs will be achieved with a bare speaker. Please recall that the speakers do not arc from ionizing the air, but rather from the diaphragm's touching the stator. In any case, we do not generate high enough drive voltages to ionize the air. If we did, good insulation would help.

Finally, the resonant frequency of the speaker is not important as long as it is well below the crossover frequency. The diaphragm will stretch out of shape with a stretcher or will shrink only to a certain tightness when you use heat. In addition, the resonant frequency has been limited to below 200Hz in all cases I have checked (mine is 44Hz). You can, therefore, ignore resonance in this hybrid system. On the other hand, high and uniform diaphragm tensions are necessary to achieve stability and high polarizing voltages, which lead to high SPLs. We want all the tension we can get and can ignore its effects on resonance.

I hope these explanations have been helpful.

INDUCTOR DEFENSE

Two letters in *SB* 1/83 remarked on my custom wound inductor article (3/82, p. 20). While I am gratified to have Wilfred Harms (p. 41) among my readers, I must take exception to his comments. First, I have not overlooked the problem of using nominal speaker impedance values for crossover design. In my article, I specifically acknowledge that speaker impedance at crossover "often is different from the speaker's nominal value of 8 or 4 ohms. Consult the manufacturer's impedance graph of the speaker, or for more accurate results, test the particular driver to determine its actual characteristics."

The avowed purpose of the article is to enable home constructors to assemble inductors to custom sizes and DC resistances, allowing them to tailor a network to real-world conditions. For those without test equipment, being able to cut a piece of wire to length, roll it up on a core and be reasonably certain of its inductance is quite handy. Those with more elaborate test equipment can check the inductor's behavior *in situ* and adjust its performance accordingly.

On Mr. Harms's other point, it is true that a correctly sized ferrous-cored inductor will perform satisfactorily. Because its magnetic field is concentrated, it requires less wire with less DC resistance loss than its air-cored counterpart. As tape-head designers attest, however, despite ferrous materials' superior concentration of magnetic force, they do not react to applied and withdrawn magnetic fields in a linear way.

Consider residual magnetism. This property is roughly the magnetic analog of dielectric absorption in a capacitor. Magnetic energy is stored in the ferrous core rather than draining to zero level during each audio cycle, much as the electrons are not all "released" in each cycle of a capacitor.

Hysteresis and coercivity are other ferrous nonlinearities that air cores do not share. These are described by their familiar loop shapes when graphed. The effects would be less important if audio were steady-state tones, for the nonlinearities within each cycle would then balance out. Since music is composed of complex wave forms, ferrous nonlinearities do affect even low-level signals.

Mr. Harms takes exception to my citing a *caveat* in Richard H. Small's work on constant voltage networks (*JAES*, January 1971). To paraphrase, Small suggests you take into account the voice coil inductance when sizing crossover networks. This remains good advice, regardless of Mr. Harms's later work in *SB* 2/82 (p. 18), which goes on to say that you must take more into account when designing real networks than an idealized resistive load.

Inductors in a crossover network operating into an inductive voice coil will require series treatment for accurate analysis. Mr. Harms's work is an *addition* to the literature on crossovers, not a replacement.

Another letter by Scott Ellis (p. 43) comments on difficulties in winding coils. I measure out the wire with a tape measure and helper who walks away with one end of the wire. I then leave the wire temporarily loose on the ground and proceed to wind it up on the core. The larger coils (3mH or so) can tangle, but not hopelessly. You can remove the kinks by bending the wire exactly opposite to the way in which it was originally bent.

Mr. Ellis's problems with understanding the use of the time constant, T_o, apparently stem from his assumption that it is the inverse of the frequency. This is not so. The time constant is defined as $159,155/f_o$. This means that the time constant for a given frequency is 159,155 times the inverse of the frequency.

Mr. Ellis also refers to "Coyle's formulas." The formulas are not mine: I adapted them from A. N. Thiele's work in the *JAES* (June 1976). In addition, I should have cited the source of my table listing wire gauges. It is *Modern Physics*, by Dull, Metcalfe and Williams, an old high school text of mine.

Mr. Ellis's observation that the DC resistance of a commercial coil varies from that of one wound according to Thiele's model is probably best explained by the difference in physical geometry between the two coils.

Finally, since I wrote my article, in which I cited Baekgaard's 6dB series formula with a Zeta, or fudge factor, of $\sqrt{2/3}$, I have come to prefer a Zeta of 0.5. The equation now becomes

$$L \text{ (in microhenrys)} = (Z)(T_o)/0.5$$

and

$$C \text{ (in microfarads)} = (T_o)(0.5)/Z.$$

This yields a steeper attenuation in the most audible octave nearest the crossover, even though the ultimate performance is the same. The joy of the 6dB series crossover is that it is linear in amplitude and minimum phase, and it is totally forgiving in coil and capacitance value inaccuracies. An off-value coil or cap merely changes the crossover frequency or slope, but the acoustic sum remains constant.

This means if you have limited test equipment and cannot accurately balance four coils for a 12dB/octave crossover, you can at least wind one coil for a 6dB series. The price of good amplitude, phase and ease of construction is the 6dB series network's modest rate of attenuation. I have not found this to be as much of a problem as some of the literature might indicate. With the advent of wide bandwidth drivers of high efficiency (by JBL and Dynaudio, among others), we are seeing a resurgence of interest in 6dB crossovers. This is appropriate, I think.

Daniel Patrick Coyle
Grants, NM 87020

LETTER WRITERS AHOY . . .

We need your cooperation in the matter of your welcome letters to authors and other readers. Please enclose a stamped and addressed envelope if you expect a reply. If the author/reader lives outside the USA, please include two International Postal Reply coupons (available at your post office) instead of stamps on your envelope.

In questioning authors, please leave room in your letter for replies which should relate to the article, be framed clearly, and written legibly. Please do not ask for design advice or for equipment evaluations.

Letters to authors or other readers cannot be acknowledged, unfortunately. Any letter which does not comply with the requests above will not be answered.

YOUR EMINENCE

I'm writing in response to Max Knittel's letter on dual coil Q_S (*SB* 2/82, p. 38). I also purchased a 12-inch dual VC Eminence from McGee. Unlike Mr. Knittel, however, I had completely satisfactory results with my unit. Careful tests gave the following parameters with one VC open:

- $f_s = 14Hz$
- $Q_{TS} = 0.44$
- $V_{AS} = 22.3 ft^3$

A 6.1 ft³ enclosure (not corrected for braces, but stuffed with Dacron) yielded a system Q of 0.955 and a system resonance of 27.5Hz.

Perhaps Mr. Knittel's woofer suffers from a malady such as cone drift—i.e., the cone is heavy. Q_{TS} is, of course, dependent on more than just magnet weight, and the manufacturer could have lowered the Q cheaply by using more layers of VC.

Roy C. Koeppe
Tulare, CA 93274

CLASSIC RETURNS

In my review of Merhaut's *Theory of Electroacoustics* (*SB* 4/82, p. 36), I mentioned in passing Prof. Frederick V. Hunt's out-of-print classic, *Electroacoustics*. I have recently learned that the Acoustical Society of America (ASA) reprinted this volume in a paperback edition just a few months ago and that you can order it for $15 per single copy or $12.50 each for five copies or more (attention, audio clubs!). Write to the ASA at 335 East 45th Street, New York, NY 10017.

Scott B. Marovich
Palo Alto, CA 94303

EXPONENTIAL HORN ADVICE WANTED

I would like to see an article on building exponential horns, including expansion rate, throat dimensions for 10, 12, 15 and 18″ low-frequency woofers, mouth areas, and axis length. I am familiar with Dinsdale's articles in *Wireless World*, but 3″ or oval-shaped speakers are not appropriate for my sound-reinforcement needs. I also read your article on Voigt's tractrix horn curve and expansion rates (*SB* 2/81, p. 9), but got lost in all the formulas. Consequently, exponential horns seem to be my best option, but I need more information about constructing them.

Also, please do not assume that all speaker builders use Brand X or McGee Radio Products, for example. Some, including myself, do use Altec, JBL and ElectroVoice products and would not consider using anything else.

C. Darrell Freeman
Birmingham, AL 35212

I don't know why Mr. Freeman thinks we

Regards the bad odor of series
Crossovers:

8/20/88

11502 Ice Caves Rd.

Grants N.M. 87020

Dear Mr. Weems,

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

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

In "Active and Passive Filters as Loudspeaker Crossover Networks",(JAES, 6/71), Robert Ashley and Allan Kaminsky discuss series crossovers. They mention 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 the notation of zeta for lower case "a" in Baerkgaard and omega for the wave number, which Baerkgaard normalizes to one. See equation 1.)

The value of "a" or zeta may be thought of as the inverse of a damping factor. It is in the selection of this value that Ashley and Baerkgaard differ. If the damping factor is changed then the behavior near resonance is affected. One may choose a value to optimise 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 zeta is set to 0.5 then the filter behaves as a quasi-second order filter, by which is meant slightly steeper roll off rates at crossover. Baerkgaard choose the root of 2/3 to optimise 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 of twiddling L and C: 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 what 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.

P.S. I thought your book was top notch,

Sincerely,

Daniel Coyle

Daniel Coyle

"assume all speaker builders use Brand X . . ." Authors choose their drivers with no direction from us, and several authors have chosen to use the brands Mr. Freeman mentions. I would be interested to know what reasons Mr. Freeman has for ". . . not using anything . . ." other than Altec, JBL or ElectroVoice.—Ed.

MAGNET MALADY

Recently I unmounted the drivers in my speaker system to check the connection to a dead driver. To my surprise, I found the driver's magnet plates covered with a milky white dust similar to the corrosion found on car battery connectors. I checked the other speaker, and it was the same. The woofers were in the worst shape, as their chassis and magnet plates were completely covered with the chalk-like dust.

I have a three-way transmission line system based on *TAA* plans from some years back. I also built a pair of fully enclosed three-way speakers for a friend using the same Philips drivers. He is strong on chemistry, and when I called him to tell him about the dust, he suggested I clean the drivers with baking soda and a small amount of water. It did the trick. I dried them thoroughly, then spray painted them to protect them from future corrosion.

For curiosity's sake, we checked my friend's system, which looked like new inside and out. We can't understand why I had this problem and he didn't. We know the humidity is high, but both our homes are air conditioned during the summer, which dries out the air, and we can't think of any chemical reaction that might have occurred at the frames. The only difference between our systems is that his speakers are fully enclosed and mine vented; his are damped with fiberglass and mine with dacron. Maybe dacron holds humidity and fiberglass doesn't?

Does anyone have any ideas or suggestions regarding the origin of this corrosion? If so, I would love to hear from you.

Rafael Lopez
Miami, FL 33125

SUSPENSION PROBLEMS

I have been using several of the Jordan 50mm modules and have experienced some suspension problems that seem to be fairly common. The modules are not burned in any way. I would love to have them back in good working order, but returning them to England for repairs costs more than buying new ones. Does anyone know what the problem is and how to repair it?

Daryl Jensen
Weimar, CA 95736

SUMMING IT UP

Controversy seems to surround the issue of summing the bass input to a subwoofer below about 100Hz. Those in favor of doing so cite the cancellation of vertical signals—such as rumble and warp—from the cartridge as the major advantage. On the negative side, opponents note the loss of perceivable stereo separation, despite the low frequencies involved. Whatever the merits of summing, another practical problem is the use of a lower crossover frequency than might otherwise be desirable, just to avoid the loss of perceivable stereo separation that everyone seems to concede (i.e., generally above about 100Hz).

I had often wondered if anyone had ever tried a hybrid approach, using a higher crossover frequency to ease the burden on the midrange unit, perhaps 400Hz, while summing only the signals below, say, 80Hz. It seemed to me that you could do this by using an 80Hz filter on each bass channel, summing the low-pass result, and then summing that summed signal with the separate high-pass signals. In that arrangement, you would choose a low-pass crossover frequency that would cancel the cartridge's low-frequency vertical output (what is the highest summing frequency necessary to get rid of the garbage reasonably?), selecting the overall subwoofer crossover frequency on the basis of the subwoofer and midrange drivers' frequency-handling abilities.

This, I thought, would allow the best of both worlds, provided you were willing to accept the loss of stereo separation below whatever lower crossover frequency you chose.

Then, just as I was preparing this letter, I came across an *Audio* (August 1983) review of Audio Control's Richter Scale crossover and equalizer. Audio Control seems to have incorporated this concept into that unit, as it includes optional bass summing below 200Hz at a 12dB slope, with either a 100Hz or 800Hz crossover frequency. Although my hybrid idea has been incorporated in a commercially available unit, I am curious about the choice of the 200Hz starting point. That seems high relative to the likely loss of stereo separation. What do you think?

Anthony Cary
San Francisco, CA 94111

QUAD TIPS

If you own a pair of QUAD ESLs, you have, in my opinion, possibly the finest sounding loudspeaker money can buy. If you have a private room for listening to music, you are advised to remove the cabinet grilles (front and rear) to obtain a still better transparency.

A monophonic amplifier for each channel should be put as near as possible to the speakers. When using biamplification for deeper bass, suspend the units about one foot above the floor to avoid floor reflection. This reflection, although enhancing bass, produces a resonance in the 240Hz region. With crossovers at 100Hz in this mode, you may remove the additional filtering from the audio transformer introduced with serial number 16,800 (March 1966). This was devised to protect the treble unit. The filter includes two 560pF capacitors from pin 8 to 9 and the 270k resistor from pin 8 to 13, jumping pin 8 to 9.

With your amplifiers positioned as near as possible to the ESL, you have the option to take a feedback voltage to the amplifiers from pin 3 of each loudspeaker audio transformer. Use a 1.5 to 2M ohm and a 1W resistor from pin 3 of each audio transformer to a tagboard, running the signal that appears before the resistor all the way back to the amplifier's feedback point via shielded twin audio cable. This procedure will reduce the influence of both the speaker cables and the audio transformer.

If you are using valve (tube) amplifiers, use the 8 ohm instead of this 16 ohm output, which will allow a better tonal balance, given the unique impedance curve of the QUADs.

If you use a four pin phono cartridge, wire it in an inverted manner, that is, using the left channel signal lead (white) for left ground and the left channel ground lead (blue) for left signal. Use the right channel signal lead (red) for right ground and the right channel ground lead (green) for right signal. This produces a more coherent acoustic phase with the QUADs.

If you have a large enough listening room, you may want to try placing each unit at a minimum distance of one meter from any wall.

To obtain the best possible result from your QUAD ESL, eliminate the small cabinet resonances by gluing ceramic tiles firmly to the outer face of the cabinet floor board, and substitute marble strips for the wooden curved lateral cabinet pieces.

Holbein Oliveira de Menezes
Rio de Janeiro, Brasil

JANSZEN WOOFER REPLACEMENT

I had never been satisfied with the stock woofer in my Janszen 412HP speakers because the electrostatic elements were much faster and sharper sounding than the woofer. When I noticed an ad in *Audio* for polypropylene woofers from Speakerworks, I investigated and found an impressive substitute for the stock woofer—with the same size, holes, and sensitivity and almost the same impedance. For $34.50 plus shipping, I couldn't go wrong.

The only problem I have found is that the speaker cabinet volume is a bit too large for the new woofer. You should reduce the volume to about 2 ft^3 with dense foam rubber, which you can find in an upholstery supply store. (Fiberglass will not perform the same function.)

I'm still experimenting with this, listening and checking the proper resonance point (around 40Hz) with a test record. I have flat upper and middle bass (no rises or heaviness), good lower bass (flat down to around 50Hz), and tight, clean, clear, fast lows. A by-product of the flatness and speed of the new woofer is an added sense of spaciousness. For a minimal expense, I have derived the maximum benefit from my speakers.

The woofer (and a polypropylene mid-range driver) are available from Speakerworks, 1910 Seneca Lane, Mt. Prospect, IL 60056. Incidentally, the designer of the woofer (and also the mid, I believe) is Chuck McShane, former director of loudspeaker R & D at Acoustic Research.

Tom Fonte
Torrance, CA 90523

Foam rubber might work well for reader Fonte, but why not use clay or cement bricks? The rubber is more expensive and might introduce a variable that is difficult to measure.—Ed.

NEW TL WOOFERS

The Madisound ten inch polypropylene woofer has an impedance rise at the crossover frequency (700Hz in the TL-10), which prevents the crossover from rolling off as predicted. An impedance compensation network solves this problem. This should be an 8 ohm resistor (10W) in series with a 25 micro F cap. Connect this network in parallel with the woofer (*i.e.*, across the woofer terminals). The easiest location is on the crossover.

The Audax ten inch woofer may also benefit from impedance compensation, though the problem is not as great as with the Madisound. For the Audax, the correct network is an 8 ohm resistor in series with a 10 micro F cap. Builders interested in using impedance compensation networks with other systems should observe these guidelines: 1. The resistor's value should equal the driver's impedance. 2. The value of the capacitor will be between 10 and 50 micro F. Fine tune it with an AC voltmeter connected across the woofer terminals and a sine-wave generator connected to your amplifier. Select the cap that makes the crossover roll off at the predicted rate. Use film type caps or electrolytics bypassed with films. This type of network compensates for impedance rise at the upper end of the driver's operating range and *not* at its resonance frequency.

In regard to the fiberglass lining behind the woofer: if you use the Madisound ten inch polypropylene, line each surface behind the woofer with a double layer of the Radio Shack fiberglass. This means the bottom, back, underside of the step, and both sides of the enclosure.

Finally, do not use the fixed resistive attenuators (R1 and R2) with the midrange and tweeters if you use the Madisound woofer, because it is more efficient. Builders who have encountered a midrange deficiency should remove the resistors; the L-pads provide ample attenuation.

Gary Galo
Potsdam, NY 13676

Tools, Tips & Techniques

Continued from page 30

tribution of the left speaker is now a decibel or two greater than that of the right (*Fig. 4*). This should help compensate for the delay in the left speaker's direct energy—about 2msec in this instance.

The net subjective effect is quite good. Center stage remains centered between the two speakers, even for extreme off-center positions. Of course, the phantom center image cannot be made perfectly stable because a loudspeaker's directivity is so frequency dependent.

In general, the less directional a speaker is, the greater θ must be for optimum operation. An 8-inch, two-way system produced the best effect with θ equal to 70°, but a little subjective testing will allow you to adjust θ. The actual angle is not too critical.

Although I have limited my discussion to home stereo systems, this concept is applicable to mobile installations. With a little care and creativity, you should be able to produce a balanced stereo image between passengers in the vehicle's front seat.

Bob Kral
Berwyn, IL 60402

Gas Guidelines

In response to the USA-61 kit report (*SB* 1/83, p. 30), I cannot help wondering whether the author's trouble with the Dalesford D-153 (separated cone and surround) stemmed from not allowing enough time for the gases from the underseal to dissipate. When using such substances, you should always let cabinets air dry thoroughly for at least three to four days to prevent gases from damaging cone-surround bonding agents. A little patience early might save much agony later.

Eric Pitschmann
Akron OH 44304

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DBP-6MC RESISTIVE LOADING KIT....................34.95
Allows adjusting load resistance from 10 to 200 Ohms for moving coil cartridges. Gold plated phono plugs in both kits.
DBP-8 SPEAKER WIRE 12 ga., in 3, 6, 9 meter and bulk.
DBP-9AU BANANA PLUGS Eight gold plated, solderless..14.95
DBP-9H BANANA HANDLES Four red, four blk............5.50
DBP-9P GOLD PLATED DUAL BANANA PLUGS 2 pk....14.95
DBP-9J GOLD PLATED DUAL BANANA JACKS 2 pk....15.95
DBP-10 PHONO ALIGNMENT PROTRACTOR...........21.95
Allows adjusting the lateral tracking error of a mounted cartridge to within ¼ of one degree. Non-technical instructions & case included.
DBP-12 AUDIO CABLE 10 meter (33 ft)..................65.95
Low capacitance (400pF) stereo interconnect cable, terminated with rugged gold plated phono connectors. Custom lengths available.
DBP-13J GOLD PLATED PHONO JACKS (¼") 8 pk.....12.95
DBP-13JR GOLD PLATED PHONO JACKS (⅜") 8 pk.....15.95
DBP-13P GOLD PLATED PHONO PLUGS 8 pk............7.95
DBP-13PM GOLD PLATED PHONO PLUGS, 4 red, 4 blk..21.95
DBP-14 GOLD PLATED SPADE LUGS 8 pk...............5.95
DBP-CK CRAMOLIN AUDIO KIT contact treatment........14.95
ELECTRONIC CROSSOVERS...6, 12, 18, 24dB..........Inquire
At your dealer or direct. Orders under $45, add $2.50 Handling.

DB SYSTEMS
Main St., Rindge, NH 03461 (603)899-5121

CLUBS

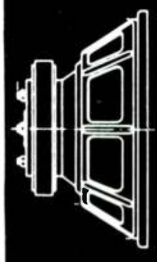
Space in this section is available to audio clubs and societies everywhere free of charge to aid the work of the organization. Copy must be provided by a designated officer of the club or society who will be responsible for keeping it current. Send notices marked Audio Clubs in care of the magazine.

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., Palo Alto, CA 94303 for newsletter.

POLYPROPYLENE MYLAR CAPS and AIR-CORE COILS
made to order
CHOUDHRY ELECTRONICS
PO Box 1743
Santa Monica, CA 90406
(213) 450-5113

SARASOTA AUDIOPHILES interested in forming a club—write: Mark Woodruff, 5700 N. Tamiami, Box 539, Sarasota, FL 33580.

AUDIOPHILES INTERESTED IN FORMING an audio club in the Washington, D.C. area please contact: Joseph Kmetz, 9861 Goodluck Rd., Apt. #10, Lanham, Maryland 20706 or call days (301) 794-7296, eves. (301) 585-3186.



Madisound Speaker Components
8982 Table Bluff Road
Box 4283
Madison, Wisconsin 53711
(608) 767-2673
Audax, Dynaudio, KEF, Philips Loudspeakers

SERIOUS AUDIOPHILES interested in a central Colorado group (Denver, Boulder, Ft. Collins, Greeley area) contact James S. Upton, 2631 17th Ave., Greeley, CO 80631.

MINNESOTA AUDIO SOCIETY. Monthly programs, newsletter, special events, yearly equipment sale. Write: PO Box 3341, Traffic Station, Minneapolis, MN 55402.

THE VANCOUVER AUDIO SOCIETY publishes a monthly newsletter with technical articles, humor and news of interest to those who share our disease. We have 50 members and meet twice every month. Dues are $15/year which includes 12 newsletters ($15 US outside Canada). Call (604) 874-3225 or write Dave Mann, VAS, Box 4265, Vancouver, BC, Canada V6B 3Z7. We would like to be put on your mailing list.

SOUTHEASTERN MICHIGAN WOOFER AND TWEETER MARCHING SOCIETY (SMWTMS). Detroit area audio construction club. Meetings every two months featuring serious lectures, recording studio visits, design analyses, digital audio, AB listening tests, equipment clinics, annual picnic and audio fun. Club publication, *LC, The SMWTMS Network*, journals the club's activities and members' thoughts on audio. To join or subscribe call (313) 544-8453 or write SMWTMS, PO Box 1464, Berkley, MI 48072-0464.

NEW JERSEY AUDIO CLUB. Emphasis on construction and mods, electronics and speakers. Beginners, students, musicians, everyone welcome! Contact Bob Young, (201) 381-6269 or Kevin Carter, (201) 277-3231, 14 Irving Pl., Summit, NJ 07901.

AUDIOPHILES IN THE NEW JERSEY and Staten Island areas interested in forming a club please contact Joe at PO Box 1588, Staten Island, NY 10314, (212) 698-2280.

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 how's and why's of audio. Each meeting consists of a lively discussion topic and equipment demonstrations. For information on meeting dates and location contact Bob Keaulani at 1902 South King Street, Honolulu, HI 96826, (808) 941-1060.

CONNECTICUT AUDIO SOCIETY WANTED. Serious audiophiles in Conn., or Putnam or Dutchess Co., NY, contact John J. McBride, 33 Perry Dr., New Milford, CT 06776, (203) 355-2032.

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 quarterly journal plus participation in meetings and lectures. Membership fee is $10 per year. For more information, send SASE to: CAS, 4506 Osceola St., Denver, CO 80212.

THE NEW YORK AUDIO SOCIETY meets monthly with prominent guest speakers, discussions and demonstrations of the latest equipment. Its $20 annual membership dues include a subscription to S/N, the society's quarterly publication. For a free invitation to our next meeting, call (212) 544-1222, (212) 289-2788 or (201) 647-2788 or write us at PO Box 125, Whitestone, NY 11357.

SAINT LOUIS AUDIO SOCIETY meets monthly for discussion and equipment audition. For information sheet send a stamped, self-addressed envelope to SLAS, 7435 Cornell, Saint Louis, MO 63130.

A CLUB FOR FM AND TV DXers, offering antenna, equipment and technique discussions, plus updates from FCC on new station data. Monthly publication "VHF—UHF Digest." Annual convention in August. For more info: Worldwide TV-FM DX Association, PO Box 97, Calumet City, IL 60409.

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

THE BOSTON AUDIO SOCIETY INVITES you to join and receive the monthly B.A.S. SPEAKER with reviews, debates, scientific analyses, summaries of lectures by major engineers. The BAS was the first to publish info on TIM, effects of capacitors, tonearm damping, tuner IM distortion, Holman's and Carver's designs, etc. Sample issue $1, subscription, $16/yr. P.O. Box 7, Boston, MA 02215.

PRIVATE FOR SALE

KEF drivers: two B110's, two T-27's. Also two crossovers with 104ab nominal values. Caps are all Mylar and metallized polyester (except for one n-p). Best offer. Steve Williamson, 33420 Michigan Ave., #38, Wayne, MI 48184.

Altec speakers—custom mirror-image pair, model 19 monitors, $800; components include Altec 416-8B, 802-8G, 811-B, N-801, owner built oak veneer cabinets. Single Altec 802-8G, $125; pair Transylvania Power Co. "tubes," $50. All FOB Houston, TX. Mike Sullivan, 2410 Ralph St., Houston, TX 77006, (713) 523-3373.

Jordan modules, $110/pr.; JVC ribbons, $28/pr.; JVC equalizer circuit, $11/pr.; Morel MDT-28, $35/pr.; Peerless PHT-25 tweeters, $13/pr.; SEAS H107 tweeters, $10/pr.; Becker 8" polypropylene 908A327, $32/pr. Assorted crossover components. Prices include shipping. Eric Pitschmann, Torrey Hall, Rm. 108, 282 Torrey St., Akron, OH 44304, (216) 762-8142, (216) 524-6684 from Dec. 18-Jan. 16.

JOIN AN AUDIO CLUB

Expand your horizons. Improve your system. Learn about the latest equipment and techniques. Share viewpoints and experiences. Don't miss out on the fun and value of belonging to an audio society.

Typical activities include:

- ***Guest Speakers.*** Here's your chance to listen to and meet "superstars" of circuit design, prominent manufacturers, acoustical consultants, and recording engineers.
- ***Tours.*** Get a behind the scenes look at the equipment and talk with the people who operate it at local TV and radio stations, universities, research labs, recording studios, and factories.
- ***Newsletters.*** These publications are often of high technical quality and are full of worthwhile information even if you don't attend many meetings. Ads and reviews help you find the right equipment, the latest records, and the dealers who carry them.
- ***Evaluation and Testing.*** Frequently clubs sponsor clinics so you can bring in your equipment for checkups on test equipment most individuals don't own.
- ***Group Buying.*** This can be an effective way to obtain obscure items from abroad, including audiophile disks.

No club in your area?

Start one—with a **free classified ad** in *Audio Amateur* or *Speaker Builder.*

For more information, see the club listings in the Classified Ads of this issue.

Classified Advertising

Jordan crossovers, new pair, $50; Bearcat 220 20-channel scanner, $150; Rolex Daytona Cosmograph, $390; 4 Jordan modules (used 5 months), $45 each; Commodore NAV 60 flight computer & manual, $45; KEF B139's, $110. Russ Bleakley, PO Drawer H, Foresthill, CA 95631, (916) 367-2292.

Peerless KJ20 DMR 2″ dome mids, $38/pr. PHT 19 tweeters, $19/pr. KEF B139 flat woofers, $175/pr., with original boxes, hardware. SEAS H204 3″ dome mids, $55/pr. Audax MHD 17HR37TSM 6½″ cone mids, $62/pr. HD13D37 1½″ dome mids, $42/pr. Steve Fritz, 5227 Calle Cristobal, Santa Barbara, CA 93111, (805) 964-0245 (weekends only).

Air 2.2 amp, new $1099; Belles I amp, black, new $549; Belles I preamp, black, new $439; Robertson 4010 amp, new $699; Denon DRM3 cassett $349; pair new Dynaudio 12″ plastic cone woofers, one has slightly damaged surrounds, $110; two pair Strathearn ribbons in beautiful oak dipolar panels, new $900, sell for $600. All above include shipping. Mike, 1919 S. 19th St., LaCrosse, WI 54601, (608) 781-2110.

Pyramid ribbon tweeters $600; RH Labs SB-2B oak woofers, pair $475; ARC D-305B $1,800; ARC ST-70-C3 $175; Electrocompaniet Ampliwire II $900. David Shreve, 319 Concord, #7, El Segundo, CA 90245, (213) 322-4623 evenings.

Advertising Index

Marantz 120 tuner, $225; Jensen metered MOSFET 150, $500; Jensen Superfet PAT-5, $300; Thorens TD125 MKII, $250; custom long-pivot tonearm (superb), $150; custom Marsh preamp, best components (polyprops, teflon, Techlab pots, rack-mount, toroidal separate power supply), $400. Wally Kowalski, PO Box 205, Sumas, WA 98295, (604) 792-6353 home, 792-0794 work.

Acoustat 2+2, $1,100; Threshold Fet 2, $800; Threshold S150, $850; Lux PD-121 with Grace 707 and F9E, $500. All in mint condition. Ron Sutherland, PO Box 741, Lawrence, KS 66044, (913) 749-0133.

Pair of English Goodman 15″ Triaxial 415 speakers, with or without walnut enclosures or plans from Goodman. Have had no high volume operation. Original owner. Replaced with KEFs. Best offer. Genuine walnut cabinets. Enclosure for equipment (36″w x 32″h) $50; speakers (18 x 18 x 32h) $35/pr.; TV (44w x 32h) $50. All matching. $120 for all or best offer. 3001 19th Ave. SE, Rochester, MN 55904, (507) 282-9966.

One JBL D-130, 15″ driver. Extended range model. 16Ω impedance. Make offer. Split shipping costs. Steve Williamson, 33420 Michigan Ave., #38, Wayne, MI 48184.

TV Sound Take-Off. *Audio Amateur* 2/78 built from Old Colony kit. Never used. $20 plus shipping. Robert Friend, 932 Larrabee St., Los Angeles, CA 90069.

Ivie 10A and 20B octave analyzer and pink noise generator. Less than two years old. Mint. $550. (2) EV SP12, factory sealed, $100 each. Gerald Ognibene, 901 Charity St., Abbeville, LA 70510, (318) 893-0174.

Magnaplanar Tympani III tweeter/midrange (pair). Marantz 15 power amp (2x60W). All in good working condition. Make offer. Rich Davidson, 501 Paige Loop, Los Alamos, NM 87544.

JR-149 loudspeakers (rosewood), $325 or trade for Hafler DH-200 or equivalent. Karl, (301) 859-5266.

Audax Satellite system (13″ x 8″ x 10″). 6½″ Bextrene, 1″ fabric dome, 18dB Mylar/air core crossover, Bitum. damping. Very quick, clean and detailed, $175/pr; Panasonic RQJ-20X (Cass/Stereo/dbx). 3 months old. List $200. sell for $110. Speakerlab 10″ 1008RS. Qt = .23, Fs = 20, Vas = 10ft³, excursion max = 1.5″. Damped cone. Perfect for subs, $40/pr. Panasonic RQ-212DKS portable tape recorder. Like new, $50. Norelco 185 Pocket Dictation Unit with 6 minisettes. New $200, sell $75. 120,000mF, 75V external filter bank, (4) 30k caps. Significant improvement to amps. $65. Call Doug at (617) 686-9691 after 6 pm.

Jordan modules, $110/pr. (2 pairs and round cabinets available); KEF T-27, $18; KEF B110, $30; Audax HD 12 x 90, $8; Audax 13D27, $20; Coles 4001 (8Ω) supertweeters, best offer; Pioneer TX 9500 tuner, best offer; Dynaco PAT-5, best offer; Transcriptor turntable with vestigal arm, best offer; Dalesford D-153, $30. Jan Waalkes, 6507 Ramsdell, Rochford, MI 49341, (616) 874-8369.

RTR electrostatic amps—350Hz crossover, $95 each; Altec line amps, $50 each; Permoflux headphones, preamp-amp, $20 each; Kronlite 50W low distortion tube amp, $175; ARC 50F-18 dual tube power amp, $325; ARC CPS-1A dual tube preamp, $195; HP650 test oscillator, $50; crossovers—JBLN500, $40; University N-2A, 2B, $10 each; Singer Spectrum Analyzer 10Hz to 40mHz, $300. Joseph R. Stephens, 41285 Crest Dr., Hemet, CA 92343, (714) 658-9575.

Jordan module speaker system using 4 per side. Enclosure is multiple materials, 1¼ inch thick, plus cork facing, $350. Excellent sound. Don, (513) 435-7155.

PRIVATE WANTED

Interested in contacting anyone working with Strathearn film diaphragm drivers. Also I.M.S. Professional MK III monitors for sale. Call (519) 837-3964 or write to T. B. Palmer-Benson, RR1, Ariss, Ontario, Canada, N0B 1B0.

Info and/or issues of *Listener's Review Finder, Arts & Music, Radio Times* (British), *Hi-Fi and Video Dealer News* (British), *Illinois Entertainer, Stereonotes* (Canadian), *Leisure Time Electronics*, and *Critical Record Review*. G. Mileon, 14 Border St., Lynn, MA 01905.

Dynaco MKIII's, Kenwood LO7MII's, Fisher R200, Scott 333B, Fisher 800C, Scott 380, Sherwood S7700, Harman Kardon TA 7000X, hi-fi buying guides and annuals from 1955-66. Trade for various speaker drivers. Steve Fritz, 5227 Calle Cristobal, Santa Barbara, CA 93111, (805) 964-0245 (weekends only).

STATEMENT OF OWNERSHIP, MANAGEMENT AND CIRCULATION (Required by 39 U.S.C. 3685) Date of Filing Sept. 28, 1983. Title of Publication: SPEAKER BUILDER. Frequency of issue: Four times a year.

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Free Distribution		
Complimentary	92	92
Total Distribution	4,819	4,638
Office use, leftover	7,180	7,362
Return from news agents	0	0
Total	12 000	12 000

I certify that the statements made by me above are correct and complete

Publication No 01997920 Edward T Dell Jr Editor

Old Colony
RECORDINGS INDEX

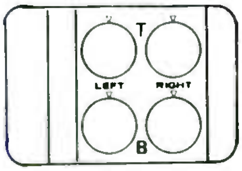
WHEN OUR COLLECTION of recordings passed the 200 mark my wife remarked to a friend that I was the only one who could find anything. When they passed the 400 mark, neither of us could. Our solution to the problem became the ***Old Colony Recordings Index System***.

Of course, if each long play disc contained only one piece of music, we would not need an index. That's how it was in the pre-1948 days of the 78s. We just arranged our heavy albums in order from Albeniz to Wieniawski. Today a single disc may contain both these and eight others in between. The spine of the average 12" package is a little over 3/16" wide and 12½" long. Even a list of composers on that spine is difficult—and the compositions outright impossible.

What is possible is a serial number for each package. A way to put it on the spine of the recording package is the heart of the Old Colony Recordings Index. To aid you in making a simple, useful index of your recorded music we offer the basic makings which you are not likely to find available locally. The system is simple, inexpensive, and flexible.

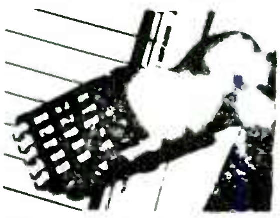
The **OLD COLONY RECORDINGS INDEX** has three main components:

COMPONENT 1



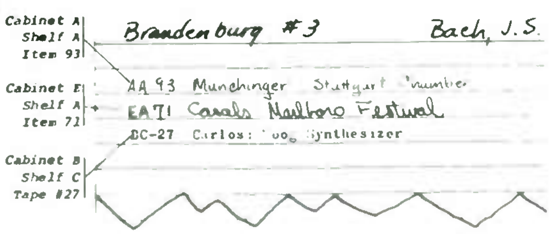
A wraparound, pressure sensitive, adhesive label for the sleeve or box of the recording. It has a place for stamping a serial number for the recording in three places.
When wrapped around the typical disc's sleeve, the number appears on front, back, and spine of the package. On multi-disc boxes or tapes the serial appears on the spine and front. The glue on the labels is permanent and does not age or deterioriate. Four small tone control knob symbols on the label may be marked to note preferred settings for listening to the indexed recording.

COMPONENT 2



Is a special adjustable, five-band rubber stamp. Its characters are turned 90 degrees so they appear vertically rather than horizontally. This makes the five character serial number easy to read on the narrow edge of the sleeve spine when it is stored on the shelf.
The first two bands of the stamp are the letters A to K (omitting I) and the last three are the numberals 0 to 9. These provide plenty of flexibility in coding. The two letters can be used to designate cabinets and their shelves; the numerals consecutive serial numbers on those shelves.

COMPONENT 3



A 3x5 (or larger if you prefer) file with guides for each composer (and some performers) arranged in alphabetical order. Each card represents one work of that composer in your recordings collection.
The name of the work is listed on the top line. The serial numbers of the discs and tapes where this work may be found are listed below with the main performers noted as well.

HIGH FIDELITY

"...The Old Colony record filing system provides what appears to be by far the best solution to date.... The convenience of the...system to the user is likely to increase arithmetically as the size of his record collection increases; so the larger the collection, the better this investment will be."

stereophile

"...We have tried several different filing systems ourselves as our tape and disc collections grew, but it has become clear to us that the only sensible way of doing it is in conjunction with a file of catalog cards. We never pursued the matter much beyond that point. The originator of The Old Colony Sound Lab system did, and the result is a filing system that seems to be foolproof, is easy to use, and is about as versatile as you wish to make it."
J. Gordon Holt, in
THE STEREOPHILE,
Winter, 4-68, p. 15
Box 49, Elwyn PA 19063

A number of users have found the system ideal for indexing slides. The stamp is used directly on the slide mount, with two letters and one numeral indicating the slide storage package and the remaining numerals the slide number in the set. A 3x5" index file is set up by category: Flower gardens" Fountains" Uncle Fred" etc.

The Old Colony starter kit includes a set of printed sheets containing some 250 composers' names (and their birth and death dates). Cut out with scissors, these fit standard ½-cut 3x5" file guides, available from any good stationer for a few cents each. A tiny dot of white glue holds the names in place on the guide while it is being covered with Scotch magic mending tape (¾") for a durable finish.

In addition to the starter kit you will need to purchase locally: 3x5 file cards, 3x5 file guides (½-cut), a file box or drawer, an inked stamp pad (cloth, not foam, permanent black ink) white glue, and Scotch magic mending tape (¾"). Your only added cost for a collection of more than 250 recordings is for labels and index cards. The system works equally well for tapes, multi-record sleeves and boxes, cassettes, and video tapes.

THE OLD COLONY RECORDINGS INDEX makes it possible to index your recordings as you acquire them, regardless of size or content. The sooner you start indexing, the easier it will be to keep up with what you own. Radio stations, university libraries, and hundreds of serious collectors are enthusiastic users of the Old Colony system.

ORDER FORM

OLD COLONY SOUND LAB
PO Box 243,
Peterborough NH 03458 USA

Ship to:

Name____________________

Address____________________

City__________ State ________ ZIP______

Quan	Item	Total price
_____	Old Colony Recordings Index starter kits at $9.95 each (250 labels, 250 composer names adjustable rubber stamp, and full instructions.	$_____
_____	Labels at $1.50 per 100 (400:$5; 1,000 $12.	_____
_____	Adjustable stamp(s) at $7.95 ea.	_____
	Total remittance	$_____

Please make check or money order payable to OLD COLONY SOUND LAB. *No C.O.D.s. MasterCharge and Visa(BAC) acceptable. All orders postpaid in the USA. Add postage for 1 lb. for starter kit to foreign address. All remittances must be in USA funds*

TEST RECORDS

PERFORMANCE TESTS WITHOUT INSTRUMENTS

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For over ten years, the original series of these high-precision test records set a standard for the audio industry. The new series sets an even higher standard. It's been revised, recut and expanded.

The new series consists of eight records for professionals and one for non-professional audiophiles.

Each record contains a complete series of easy-to-use tests to help you rapidly and accurately evaluate components and systems. Even one of these records can eliminate the need for costly, additional equipment. Each will find productive use and save you hours in the laboratory, on the production line and in field testing.

OLD COLONY SOUND LAB · OLD COLONY SOUND LAB · OLD COLONY SOUND LAB

TAKE A LOOK AT WHAT THIS ESSENTIAL TESTING SERIES CONTAINS:

STR 101 SEVEN STEPS TO BETTER LISTENING. For only $6.98, you can improve your system with CBS Laboratories' "Seven Steps to Better Listening." This high-precision test record enables you to make sure that your equipment functions properly . . . to tune your system to your ears and your room acoustics. Included is a detailed 16-page booklet by *Audio's* Edward Tatnall Canby explaining how to use the record to improve your system's performance. With the record you can perform the following "ears alone" tests: left-right identification, phasing, loudspeaker balance, tone control setting, alternate phasing, buzz and rattle elimination, lateral tracking, and vertical tracking.

STR 100 STEREOPHONIC FREQUENCY TEST RECORD. Designed for the evaluation of pickups and systems. Provides a constant amplitude characteristic below 500Hz and a constant velocity characteristic above 500Hz. Tests include: Sweep Frequency—with the sweep rate synchronized for use with a graphic level recorder; Spot Frequency—with voice announcements; Channel Separation; Wavelength Loss and Stylus Wear—to pinpoint oversize or worn-out styli, and excessive pickup tracking force; Compliance; Phasing; Vertical and Lateral Tracking: Tone Arm Resonance—to check system performance at low and subaudible frequencies and thus reveal undamped resonance which may cause equipment overloading.

STR 112 SQUARE WAVE, TRACKING AND INTERMODULATION TEST RECORD. Enables detailed study of tracking capabilities of stereophonic phonograph pickups. The square wave modulation allows a rapid appraisal of stylus-tip mass, damping, and tracking. Low frequency compliance and tracking are determined by means of 300Hz bands of progressively increasing amplitude. Intermodulation distortion measurements are made possible by graduated 200Hz intermodulation test bands. The STR 112 has been cut with vertical angle approximately 15° which is representative of current recording practice.

STR 120 WIDE RANGE PICKUP RESPONSE TEST RECORD. Makes possible the measurement of pickup response at frequencies far beyond the audible range, where elusive distortion elements can cause audible distortion. The low-frequency range includes glide-tones at twice normal level for the detection and elimination of arm resonance, loudspeaker cone and cabinet rattles. Other tests include: silent grooves for measuring rumble and surface noise characteristics; and standard level bands at 0dB for overall system S/N measurements. This record is suitable for use with a graphic level recorder to provide permanent, visible records for precise evaluation.

STR 130 RIAA FREQUENCY RESPONSE TEST RECORD. Provides RIAA frequency characteristics for the calibration of professional recording equipment and for testing the response of professional and consumer record reproduction equipment. This record is suitable for use with a graphic level recorder to provide permanent, visible records for precise evaluation. Spot frequency bands for use without automatic equipment are included.

STR 140 RIAA PINK NOISE ACOUSTICAL TEST RECORD. Designed for acoustical testing of systems and loudspeakers and for psychoacoustic tests on reproduction equipment. With the STR 140 it becomes possible to test loudspeakers in the room in which they will be used. Spot frequency tones with voice announcements facilitate the testing procedure. Continuous glide-tones in ⅓-octave bands cover the frequency range from 30 to 15,000Hz and are synchronized with a graphic level recorder.

STR 151 BROADCAST TEST RECORD. Developed especially to meet the needs of broadcast engineers, audiophiles, and other professionals seeking a convenient signal source for the testing and adjustment of all audio equipment. Tests include: phonograph pickup response and separation, speed accuracy at 33⅓ and 45 rpm, wow and flutter, rumble and hum detection, ballistic test of VU meters and many others.

STR 170 FREQUENCY RESPONSE 318 MICROSECOND TEST RECORD. Provides pickup designers and recording studios with a high-level, easily-equalized signal for frequency response and channel separation measurements. The STR 170 employs a 318 microsecond characteristic corresponding to the "test" or "flat" mode common to most disc recording equipment. Constant amplitude recording is employed in the region below 500Hz with constant velocity recording in the region above. The transition is smooth, in contrast with the STR 100 which employs a sharp breakpoint at 500Hz. The record is suitable for use with a graphic level recorder to provide permanent, visible records for precise evaluation.

SQT 1100 QUADRAPHONIC TEST RECORD. Designed for calibration, verification, and adjustment of SQ® decoding equipment. The record provides test bands for pickup measurements, for adjustment of decoder electronics and for channel identification and balance. Each band is described in terms of recorded characteristics and its intended use.

SQ is a Trademark of CBS Inc.

OLD COLONY SOUND LAB
P.O. Box 243, Peterborough, NH 03458
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Credit Card # ______________________

Expiration Date ______________________

Name ______________________

Street ______________________

City ____________ State ________ Zip ________

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Quantities are limited. Be sure to order enough for many years of use.

Send me the following test records:

_______copies (STR 101) Seven Steps to Better Listening at $6.98 ea.
_______copies (STR 100) Stereophonic Frequency at $10. ea.
_______copies (STR 112) Sq. Wave, Tracking and Intermodulation $15. ea.
_______copies (STR 120) Wide Range Pickup Response at $15 ea.
_______copies (STR 130) RIAA Frequency Response at $15 ea.
_______copies (STR 140) RIAA Pink Noise Acoustical at $15 ea.
_______copies (STR 151) Broadcast Test at $15 ea.
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