WHAT MAKES ONE DRIVER BETTER THAN ANOTHER

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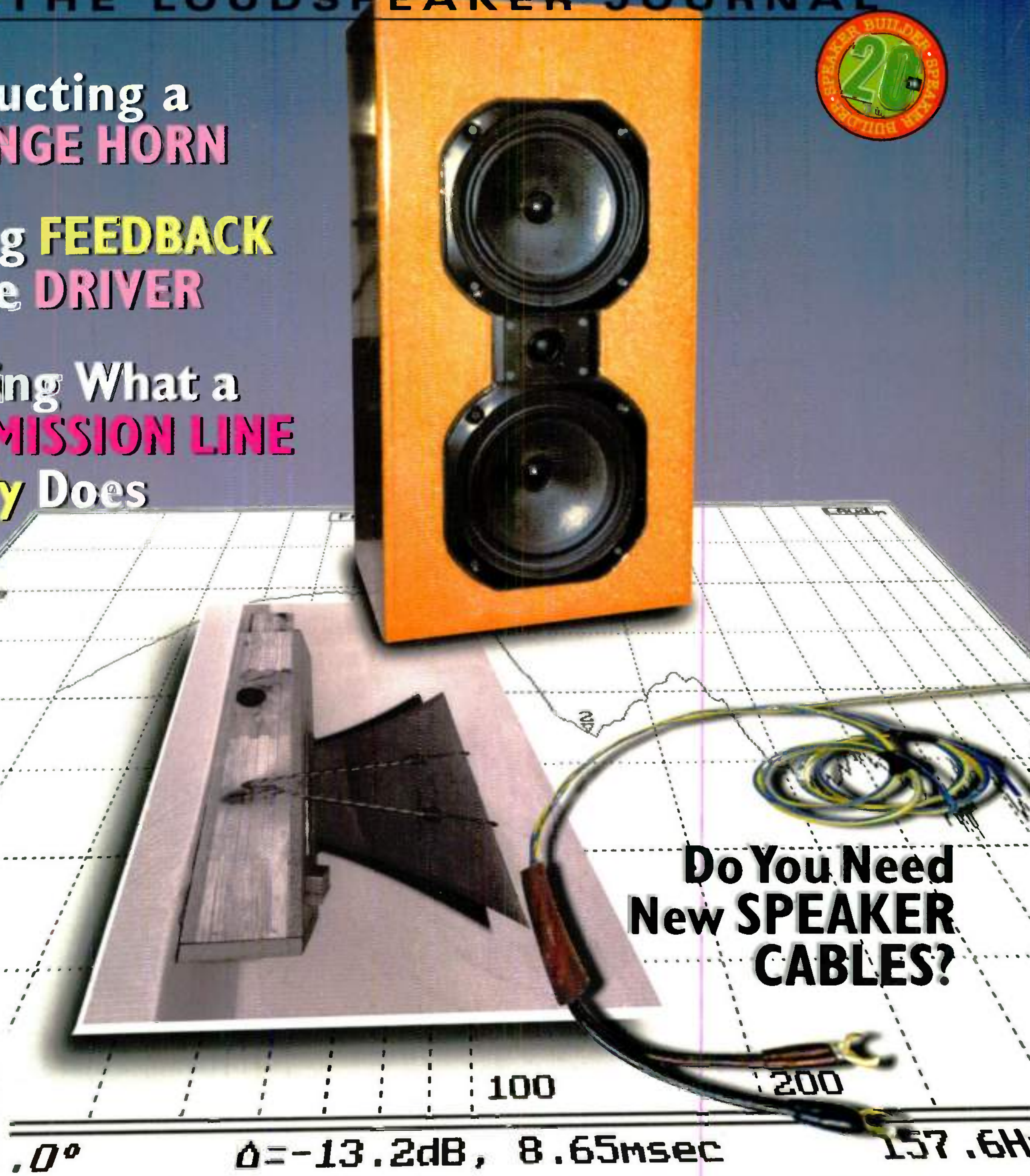
THE LOUDSPEAKER JOURNAL



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Good News

NEW GENERATION

Burnett Associates, North American distributor for RCM Akustik of Germany, announced RCM's Generation II Detonation! Series subwoofer amps with crossover. Available in both 100W/4Ω and 150W/4Ω, these amps come completely assembled and ready to mount in the back of the subwoofer speaker box. Features are remote control volume, variable Linkwitz-Riley LP crossover, adjustable bass boost and phase control, crossover bypass switch, subsonic filter, satellite filters, auto on/off/standby, toroidal transformer, a solid, one-piece chassis/heatsink, and more. Burnett Associates, LLC, PO Box 26, W. Peterborough, NH 03468, (877) 924-2383, FAX (603) 924-3392, E-mail dear2@prodigy.net, Website http://pages.prodigy.net/dear2/audio.

Reader Service #135

PARTS EXPRESS CATALOG

Parts Express has released its latest catalog, containing 284 pages of raw loudspeaker drivers for home and automotive applications as well as home theater and home automation products, alarm systems for home and car, test equipment, chemicals, telephone products, wire, connectors, instructional books and videotapes, speaker design software, cellular phone accessories, stage lighting, and pro sound equipment.

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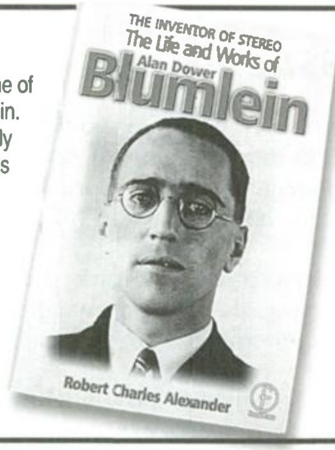
The Inventor of Stereo studies the life and works of one of Britain's most important inventors, Alan Dower Blumlein. Blumlein produced numerous patents, breaking entirely new ground in electronics and audio engineering. He is also known for the "H2S" blind bombing radar. The 448-page book provides detailed knowledge of all of his patents and the process behind them, while giving an in-depth study of his life. Included are 100 black & white photos. Available from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467, E-mail custserv@audioXpress.com.



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Reader Service #136

IN-WALL SYSTEM

Atlantic Technology's System 10 is a high-performance in-wall loudspeaker designed for home theater and music reproduction. It is configured as a D'Appolito array with a central swiveling 1″ silk dome tweeter, a front-panel high-frequency level control that can boost or cut tweeter output by 2dB, and the tweeter is flanked by two long-throw 6½″ injection-molded graphite (IMG) woofers. These drivers include powerful motors, high-excursion half-roll rubber surrounds, and magnetic shielding. Atlantic Technology, 343 Vanderbilt Ave., Norwood, MA 02062, (781) 762-6300, (781) 762-6868, Website www.atlantictechnology.com.

Reader Service #137

About This Issue

For readers who have been clamoring for more horn coverage, **Louis C. McClure, Sr.**, offers his experimental work with horns, testing throat size in the construction of a midrange version. "An Exponential Midrange Horn" (p. 10) provides a good tutorial on how to determine the horn parameters (throat area, rate of flare, and so on).

In the second part of his transmission lines analysis (p. 18), **A. Monk** takes a closer look–with measurements and graphed responses–at the many complexities affecting system response. You'll gain an appreciation of the concept of decomposition and the effects of fiber-stuffing density in a TL line.

We consider the articles we publish gems, but some shine more brightly than others. A case in point is **Hans J. Klarskov Mortensen**'s experimental work on single-ended acceleration feedback ("Acceleration Feedback Systems," p. 28).

Are you sometimes baffled by manufacturers' driver sales sheets when selecting drivers? In the second part of this "Navigating Speaker Design" series, **Mark Wheeler** provides the key to help you decode these driver parameters provided by manufacturers ("Sleuthing Driver Parameters," p. 34).

You may be asking yourself the same question that author **Jesse Knight** probes in this issue: "Do I Need a New Speaker Cable?" (p. 38). In answering this question, he shows you how to make a 60′ length of cable that addresses the problems of cable inductance and resistance.

Also in this issue, columnist **Barry Fox** reviews the recently released biography of Alan Dower Blumlein, which details his amazing inventions of stereo for recordings and movies, as well as ground radar in WWII ("Book Reviews," p. 55).

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

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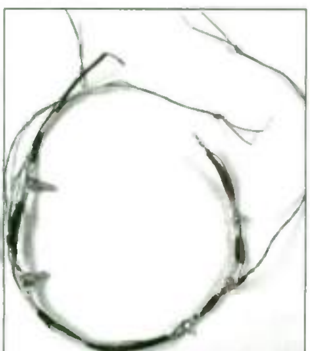
page 10

page 18



page 38

page 60

FEATURES

BOOK REVIEWS

DEPARTMENTS

IN EVERY ISSUE

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Editorial

OUT, OUT, DAMNED DOT

Nearly everyone in the North American electronics establishment seems to me content to sit, seemingly happily and permanently, between two stools. The situation concerns the unhappy choice Michael Faraday made in choosing a quantity for that relationship between two layers of metal separated by nearly any dielectric that we refer to as capacitance. We refer to these units as microfarads, which isn't exactly the way it is with capacitance. Faraday's farad is very, very large in relationship to the components we normally use.

The microfarad is actually one millionth of a farad: 10^{-6} farads. Notice the minus sign before the exponent, which means that the number is smaller than one. We represent the micro with the 12th lowercase letter of the Greek alphabet, the µ, which is equivalent of our "m." We do not use the "m" for micro because it has already been taken by the next less small mathematical quantity, the thousandth, or milli (10^{-3}). Which is why when writing about microfarads we do not designate a capacitor as being 40mF. Unless we mean a 40,000µF device. We don't generally do that since we seem to have this passionate devotion to the little Greek µ and its decimal derivatives.

At least in North America we do. We also have a positive, abhorrent aversion for the perfectly respectable mathematical representation for numbers which are in the billionths, namely the nano. No, we would rather resort to the creaky, circumambulating, ditsy business of slicing up micros with decimals. What sense decimals make when talking about billionths escapes me. It is either laziness, ignorance, the herd instinct, or perverseness, take your pick.

Any quantity of farad slices that come in billionths can be designated as nanofarads. These are units which are 10^{-9}. The whole extent of decimated farads in this range, from 0.001µF to 0.999µF, is more properly and easily designated as 1nF to 999nF. The most often used 0.1µF as a bypass for op amps is far more easily written 100nF—same number of characters, and without the Greek letter (Alt 0181, on your computer keypad).

Fortunately North American electronics practitioners have made some small progress in the last three or four decades or so since we have, at least, abandoned "µµF" as a way of designating picofarads. Pretty generally we have accepted the fact that a trillionth of a farad (10^{-12}) is properly referred to as a pico. Thank goodness we have graduated from referring to these devices as accumulators or condensers, which we did for decades (shades of the Leyden jar).

Another major problem with this lazy decimation habit is the fragility of the little dot required for decimals. Some, who are obviously afraid the little decimal will get lost in the nineteenth copy of the schematic, put ***another*** zero in the lineup just to protect the puny little thing: as in 0.001µF. If you love redundancy, help yourself. However, isn't 1nF better? I submit it is not just better, it is a ***lot*** better.

It's easier to think of these generally used designators as three sets of threes.

Larger → µ n p → Smaller
000 | 000 | 000

In this simple way, you never need use a dratted decimal—that is, unless we can all agree on some of the larger units of electrolytic capacitance for smoothing low voltages such as a millifarad: 1mF meaning 1000µF, but that might confuse everyone. However, we should be able to manage a 1mF, realizing that the lowercase "m" always means milli or thousandths. Then, of course, we could move on to 1cF, the centifarad, or 100,000µF. This change would be very helpful on power supply schematics where the space around capacitors is usually limited. I'll be happy if we can all agree to just start using the nano regularly. Perhaps this reluctance is part and parcel of the U.S. fear of that "foreign conspiracy" known as the metric system, which is standard in most of the civilized world today. But not here. It cost NASA a Mars satellite the other day.

If you argue that decimalled farads are like Gary Galo's (one of our regular contributors) favorite, the current flow issue, in that the millions of textbooks have all adopted the "mistake," and it would be just too expensive and difficult to start changing all the literature at this stage of our existence, then I do not believe the issues are comparable. We have made the change from the old micro-micro farad of the thirties and forties. We could easily begin changes in the books, catalogs, and spec sheets starting in the new century. Most of the European electronics press already has this style in place.

One historical note might be in order here. We honor Faraday and other discoverers of electronic properties by designating such quantities with a number and the capital letter of their surnames. Faraday's µF, Henry's µH and mH, Nicolo Volta's 1V or µV. Poor Georg Ohm has the misfortune to have a name whose capital is too similar to the zero, so the elders resorted to the good old Greeks again, picking the omega, Ω, to designate the ohm. When we spell these quantities, however, the proper use is lowercase, which indicates that we're talking about a quantity, not the distinguished person.

It is time to stop decimating the farad. The standard math notation for these little quantities is much more economical, rational, logical, and also neater. These magazines will continue to convert those ugly decimals to their proper designators. I hope you will spend some time thinking through the change and adopt it for yourself.—**E.T.D.**

QUANTIFICATION FOR ELECTRONICS

LESS THAN ONE

	TRILLIONTHS PICO 10^{-12}	BILLIONTHS NANO 10^{-9}	MILLIONTHS MICRO 10^{-6}	THOUSANDTHS MILLI 10^{-3}	HUNDREDTHS CENTI 10^{-2}	TENTHS DECI 10^{-1}
ampere			µA	mA		
farad	pF	nF	µF	mF		
henry			µH	mH		
hertz						
ohm						
volt			µV	mV		
watt			µW	mW		

ONE AND LARGER

ONES	TENS 10^1	HUNDREDS 10^2	KILO 10^3	MEGA 10^6	GIGA 10^9
A					
F					
H					
Hz			kHz	MHz	GHz
Ω			kΩ	MΩ	
V			kV	MV	
W			kW	MW	

Note: The discoverers of these units are lowercased referenced quantities. The letter representing them is capitalized, except for Georg Ohm whose unit is designated by the Greek letter Ω, omega.

The author constructed this large-throat horn as one of two midranges for his giant bass horn. Readers following this tutorial will learn a lot about how to determine horn parameters (throat area, rate of flare, and so on).

Part 1

An Exponential Midrange Horn

By Louis C. McClure Sr.

Recently, I needed a midrange horn capable of generating sufficient output to match a large bass horn I had built, and I intended the entire unit to be horn loaded.

The bass horn has a 12″-deep cavity in front, in which I planned to install the midrange. However, the 12″ depth limited the overall length of the midrange horn to a maximum of 8″, since I also needed to fit the driver and rear chamber, which is 4″ deep, into this cavity.

DESIGN-FREQUENCY CHOICE

I performed a frequency-response check of the bass horn, and found that it had a mass rolloff at 390Hz. Therefore, I initially chose to make the design (cutoff) frequency of the midrange horn 200Hz.

I located a 6½″ driver (the MCM 55-1585) that seemed well-suited to my needs. It is similar to the Focal Audiom 7K, with a frequency response of 100Hz-14kHz, a Q_{TS} of .55, a Q_{ES} of .58, a V_{AS} of .23ft^3, an F_S of 111Hz, and an SPL of 94dB/W/m. It also has a woven Kevlar® cone, a corrugated surround, and a phasing plug, and it sells for $49.95.

I planned to use a square-horn format, with a throat area approximately equal to the square of the maximum diameter of the surround. Since this diameter is 5.5″, the throat area is 30.25in^2.

My calculations for a horn with this throat area and a design frequency of 200Hz showed that the required length of the horn would be 12.2″, and the minimum mouth area 287in^2. Adding the 4″ minimum depth of the rear chamber to the calculated length of the horn made it 4.2″ too long to fit into the 12″-deep cavity.

PHOTO 1: Frontal view of the horn, mounted on an adjustable rear chamber.

Hence I was forced to increase the design frequency of the horn to 250Hz, which, with the throat area of 30.25in^2, resulted in a horn length of 7.8″ (*Photos 1* and *2*). The rear chamber is 4″ thick overall, which is the minimum thickness required to accommodate the driver. With these given parameters and space limitations, I proceeded to design and build this midrange horn.

To design the horn, I used the formulas as given in the *Audio Cyclopedia* for exponential horns. Technically, mine is not a horn, but a directional baffle. The difference apparently is that the throat of a baffle is equal to or larger than the cone or diaphragm of the driver. However, I shall refer to the device as a horn in this article.

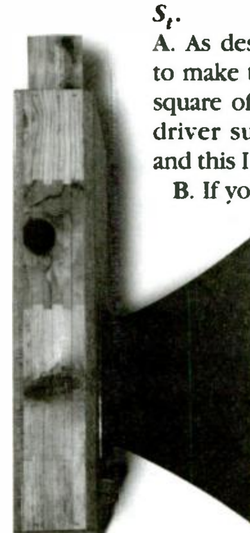

PHOTO 2: Side elevation of the horn.

The requirements for my horn were as follows:

- maximum length, throat to mouth, 8″;
- design (cutoff) frequency, 250Hz.

HORN DESIGN

Following are the steps for calculating the dimensions and contour of the horn:

1. Determine the flare constant, "M". Using the formula $M = 4\pi F/13{,}548$, (where F is the cutoff frequency), M = 0.2318866.

2. Determine the initial throat area, S_t.

A. As described previously, I chose to make the throat area equal to the square of the diameter (5.5″) of the driver surround I planned to use, and this I calculated to be 30.25in^2.

B. If you prefer to make the throat area in accordance with the formula for a restricted throat, you may use the formula $S_t = .8 \times F_S \times Q_{ES} \times V_{AS}$ for your driver. My driver's F_S of 111Hz, Q_{ES} of .58, and V_{AS} of .23ft^3 would produce a throat area of 11.85in^2.

One of my objectives

in designing and building this large-throat horn was to determine the effect of throat size on the tonal quality of the sound.

Regardless of the throat size you choose, the design procedure is the same.

3. Determine the minimum cross-sectional area of the horn mouth. To properly reproduce the design frequency of the horn, each side of the square mouth should be equal to ¼ wavelength of the design frequency. (As mentioned previously, for my design I assumed a square-horn format.) For a circular format, use the diameter of the mouth to calculate the cross-sectional area to determine design frequency.

In my case, the design frequency was 250Hz. The wavelength of 250Hz = 13,548/250 = 54.192″. One-quarter of this (54.192/4) = 13.55″ on each side, and 13.55^2 yields the minimum mouth area of $183.6in^2$.

4. Determine the cross-sectional area (S_x) of the horn from the throat to the mouth in ¾″ increments. (Alternatively, you may use ½″, 1″, or any other convenient increment.) The formula for determining S_x at a given point x inches from the throat is: $S_x = \log_e(M \times x) \times S_t$, where S_t is the initial throat area.

To use this formula, proceed as follows:

A. Use the value M (0.2318866 in my case) as determined in step 1.

B. Multiply M by the distance (in inches) from the throat. If you used ¾″ increments, you would multiply M × ¾″ (or .75″). In my case this would equal 0.2318866 × .75, or .1739.

C. Using your scientific calculator, find the natural log ($\log_e$) of .1739, which is 1.1899544.

D. Multiply 1.1899544 × the initial throat area, S_t. In my case, I used ¾″ increments, because this corresponded with the thickness of each layer of the core. This gave a throat area of $30.25in^2$, which produced an area of $35.996in^2$ at ¾″ from the throat (toward the mouth). I rounded off the area to two significant figures, giving a cross-sectional area of $36.00in^2$.

E. Continue this process for each ¾″ increment from the throat until the cross-sectional area equals or exceeds the value obtained in step 3. (In my case, at a minimum of 7.8″ from the throat, the cross-sectional area is $183.6in^2$.)

I suggest that as you calculate the

TABLE 1
CROSS-SECTIONAL AREAS

DIST. FROM THROAT (INCHES)	CROSS-SECT. AREA (INCHES²)	SQ. ROOT OF CROSS-SECT.	½ OF SQ. ROOT AREA OF CROSS-SECT. AREA
0	30.25	5.5	2.75
.75″	36.00	6.00	3.00
1.5″	42.83	6.54	3.27
2.25″	50.97	7.14	3.57
3.0″	60.65	7.79	3.89
3.75″	72.17	8.50	4.25
4.5″	85.88	9.27	4.63
5.25″	102.20	10.11	5.05
6.0″	121.61	11.03	5.51
6.75″	144.71	12.03	6.02
7.5″	172.20	13.12	6.56
8.25″	204.91	14.31	7.16
9.0″	243.83	15.61	7.81

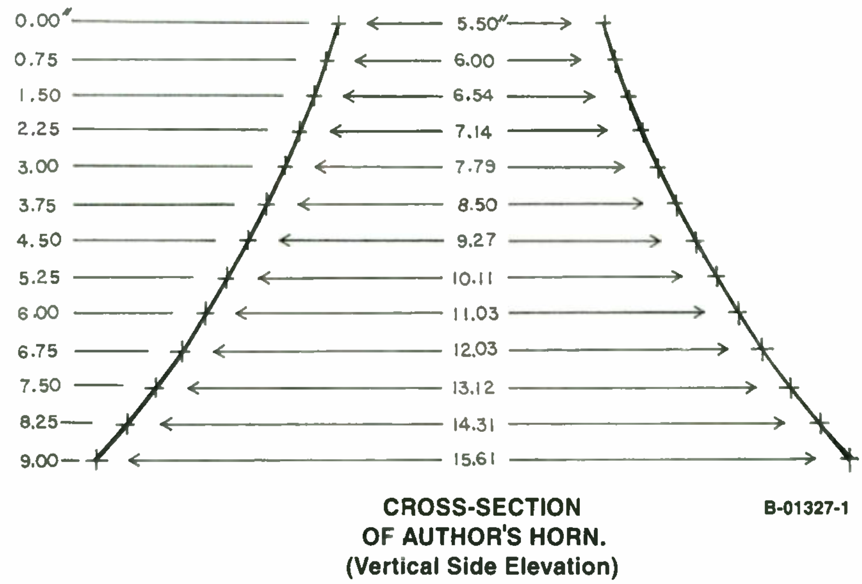

FIGURE 1: The vertical side elevation for the large-throat midrange exponential horn. I used ¾″ increments in the vertical height (not to scale). This corresponds to the thickness of ¾″ MDF, and also to the increments used to calculate the dimensions in *Table 1*. I shortened the horn to 8″ to fit into the cavity in the bass horn.

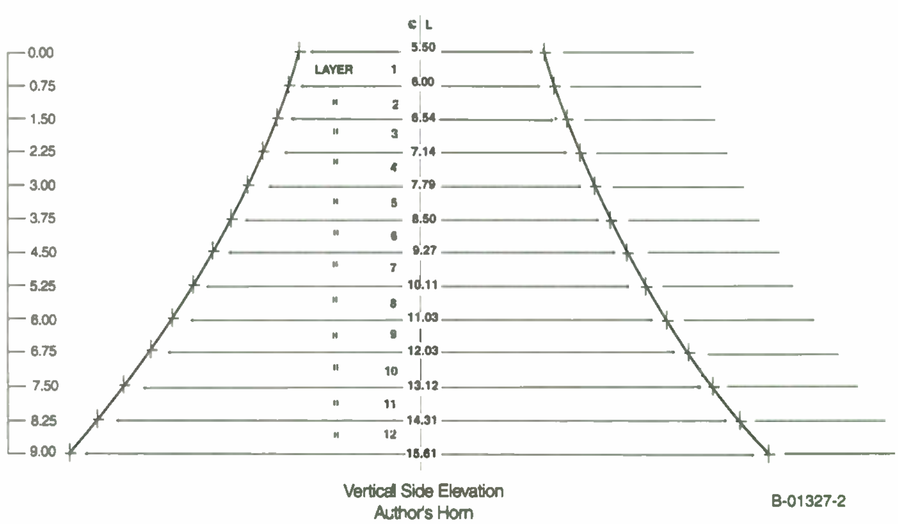

FIGURE 2: Vertical side-elevation with center line.

cross-sectional areas for your case, you also prepare a table of these similar to *Table 1*, as determined in steps 1-4. This table will be useful later when you determine the measurements and contour of the horn, as shown in *Figs. 1* and *2*.

SIDE-ELEVATION DRAWING

Figure 1 represents the vertical side elevation of the horn, which you can use to determine the overall sizes of the layers to be made, and the angles at which to cut the edges.

To plot the side elevation, or contour, of the horn, I used a large sheet of ¼″ quadrille-ruled paper. In order to make the dimensions of the horn compatible with this graph paper, I used decimals.

I began by drawing a vertical line down the center of the sheet (C/L in *Fig. 2*). Then I drew 13 horizontal lines ¾″ apart, from near the top of the sheet to the bottom. These lines represented the top and bottom surfaces of the layers of MDF material used to build the core that served as a mold for the horn.

For each of the distances shown in column 1 of *Table 1,* I measured with a pair of dividers the values shown in the fourth column (½ the square root of the cross-sectional area), and marked off these distances on the appropriate horizontal lines of my drawing to the left and right of the center line. This procedure ensures symmetry about the vertical axis. I then connected the points on the horizontal lines with straight lines, because this made it easier to measure the angles when I cut each layer of the core. This defined the contour of the core and the dimensions of each of its layers.

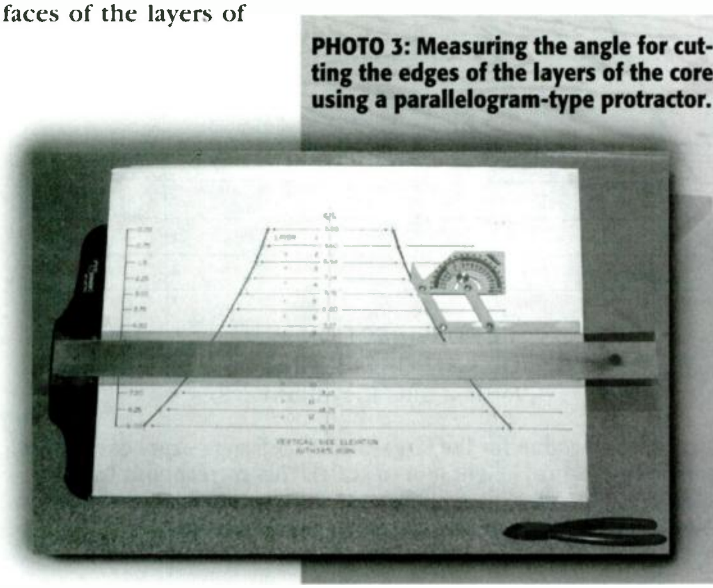

PHOTO 3: Measuring the angle for cutting the edges of the layers of the core, using a parallelogram-type protractor.

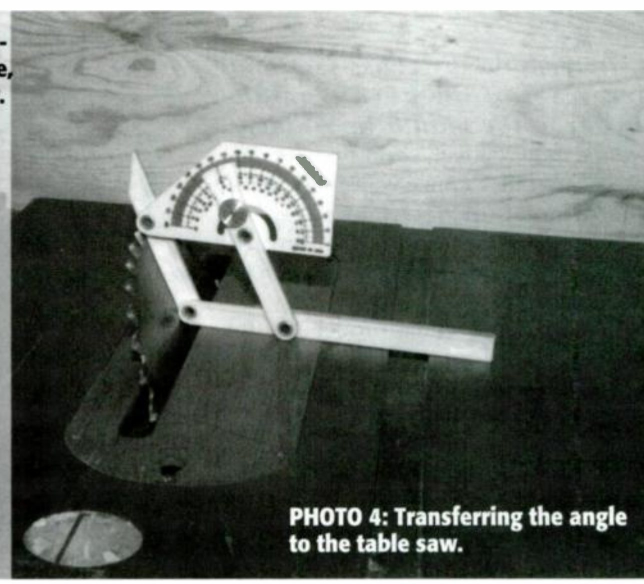

PHOTO 4: Transferring the angle to the table saw.

You will use the vertical side-elevation drawing (*Fig. 1*) later for determining the overall size of each layer and the proper angle for cutting the edges of each. I also used *Fig. 1* to make a pattern for building the gluing and assembly jig, which I'll describe later.

CONSTRUCTING THE CORE

The core is required for holding the sides of the horn in position for assembling and gluing. Refer to *Table 1* for the dimensions for cutting individual layers of the core. Use the dimensions listed in column 3 to cut out the blank layers for the core. Cut out a blank square for each layer, beginning with the largest (the mouth) and continuing up to the smallest (the throat). This also results in the most efficient use of the material.

I used ¾″ MDF for making the core, since it is uniform in thickness, flat, easy to work, and economical. It also lent itself perfectly to my initial calculations for the area of the cross-section of the horn and the vertical side-elevation drawing (*Fig. 1*), so that the ¾″ thickness of the MDF corresponded exactly with the dimensions in *Table 1* and *Fig. 1*.

After cutting out the square layers, refer again to *Fig. 1* to determine the angles at which to cut the edges of each layer. You can use a parallel-type protractor to measure the angle of the edges from the vertical side-elevation drawing (*Photo 3*). Then transfer each angle to the table saw (*Photo 4*) to cut that particular layer. Measure the overall width and angle of each layer very carefully. All four sides of each layer should be cut at the same setting of the table saw for both the layer width and the angle of the edge.

After cutting all the layers, stack and align them carefully, with the largest layer on the bottom, and the smallest on top. I fastened them together using 1½″ long nails, but you can glue them together if you prefer. This completes the construction of the core.

As an aside, you could also use the core as the basis for building a form for molding a fiberglass, plastic, or concrete horn. However, you would need to cover the outside of the core with a thin sheet of metal and seal it.

GLUING AND ASSEMBLY JIG

The gluing and assembly jig is necessary for laminating together the individual panels of the horn. The jig forms the curvature of the side panels and holds them in position while the glue is drying (*Fig. 3* and *Photos* 5 and *6*).

You can make the jig from three short pieces of 2″ × 4″ stock and one 2″ × 6″ or 2″ × 8″ piece. In discussing the vertical side-elevation drawing (*Fig. 1*), I mentioned that you could also use it as a pattern for making the jig.

I made a template using the contour of one side of *Fig. 1* as a pattern, and transferred this form to a thin sheet of aluminum, cutting out the pattern with a pair of tin snips. I made the template approximately 2″ longer than the side of the horn, from the throat to the mouth.

I placed the template on a piece of 2 × 4 material, aligning the straight edge with the straight edge of the 2 × 4, marked out the pattern on the 2 × 4, and cut out three pairs of these concave and

Swans M1 kit

Great news from Swans!

New beautifully cabinets for Swans M1 kits are available in three finishes: piano black, solid walnut and rosewood veneer. Totally irresistible!

The Swans M1 speaker system is a two-way bass-reflex design. The front baffle is very narrow with rounded edges to reduce cabinet diffraction for better clarity and imaging. The internal panel and corner reinforcement substantially reduce unwanted cabinet vibrations. A flared port is mounted on the rear baffle for smooth transition from the port to cabinet boundaries. This provides linear bass performance and absence of port noise. The heavy-duty gold plated binding posts are mounted directly on the rear panel to enable easy cable connection.

The 5-inch paper/Kevlar cone woofer has a rubber surround, cast aluminum frame and a magnetically shielded motor system. This driver utilizes a central phase plug to avoid air compression, improving frequency response and dispersion. These key features greatly contribute to the M1's clear transparent sound and effortless dynamic performance.

The tweeter is a high-tech planar isodynamic design that employs Neodymium magnets and extremely light Kapton®film, with flat aluminum conductors.

This unit provides an immediate and precise response to any transients in original signal, and gives the M1 an exceptional ability to reveal the true dynamics of instruments with a complex high frequency spectrum.

The crossover is a second order Linkwitz-Riley type resulting in an in-phase connection of the drive units. The crossover frequency between the two drivers is 3.3 kHz and only high quality polypropylene capacitors are used. Each filter has it's own dedicated board mounted on a special rubber interface to reduce vibrations and microphonic phenomenon. The filter boards are spaced inside the loudspeaker with the inductors positioned at right angles to minimize the interaction.

Swans M1 kit includes:
- 2x F5 paper/Kevlar bass-midrange drivers.
- 2x RT1C isodynamic tweeters with sealing gaskets,
- 2x dedicated tweeter crossovers,
- 2x dedicated bass-midrange crossovers,
- two flared ports,
- two pairs of heavy duty gold plated terminals.

The drawings of the cabinet shown here represent general dimensions required for optimum bass performance. Rounded corners are advisable as they improve imaging and clarity. Actual finish and appearance is a matter of personal taste.

Price: $410 delivered without cabinets;
$660 delivered with cabinets
Warranty 3 years. We accept VISA.

High sonic resolution Loudspeaker
with room friendly performance

"...explicit, easy to listen to, effortless, seamless and stunning."
Ernie Fisher
Swans M1 Speaker Systems Review
INNER EAR REPORT
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SWANS
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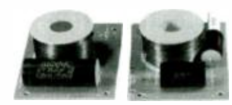
The step beyond the limits


RT1C Tweeter


F5 Bass-midrange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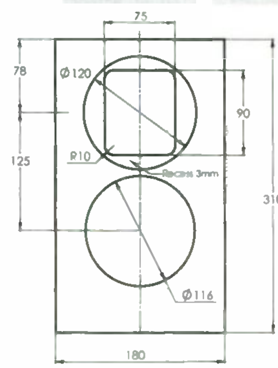
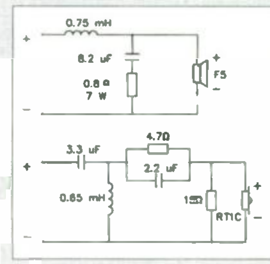
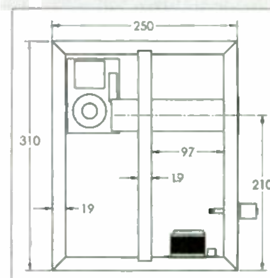
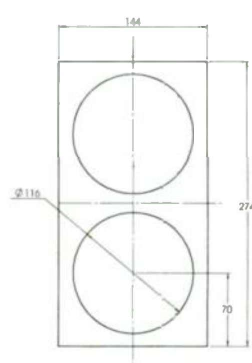


Front view

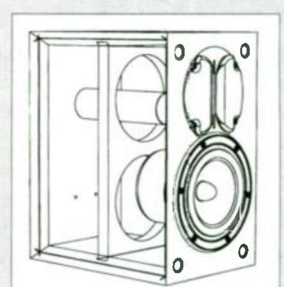




Right view of the cabinet with accessories(right side panel removed)



Internal panel



SPECIFICATIONS

Frequency response (1m,half space)	60Hz-35kHz,±2dB 55Hz-40kHz,-3dB
Sensitivity,1W/1m (100Hz-8kHz averaged)	86 dB
Nominal impedance (7.2 ohms minimum at 250 Hz)	8 ohms
Power handling	50W nominal, 90W music
Dimensions,HxWxD	310x180x250 mm

Amplifier requirements:
30W recommended minimum.

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convex formers on my band saw. I kept each set together, labeling them A, B, and C. When gluing the pieces of paneling together (see the next section), I took care to keep each matched pair of formers together, as well as aligning them end-wise.

I made a base for the formers from a piece of 2″ × 8″ material, approximately 2″ longer than the width of the horn mouth. I positioned the concave portions of the formers at right angles to and centered upon the base, and nailed each one in place. This completed the construction of the jig.

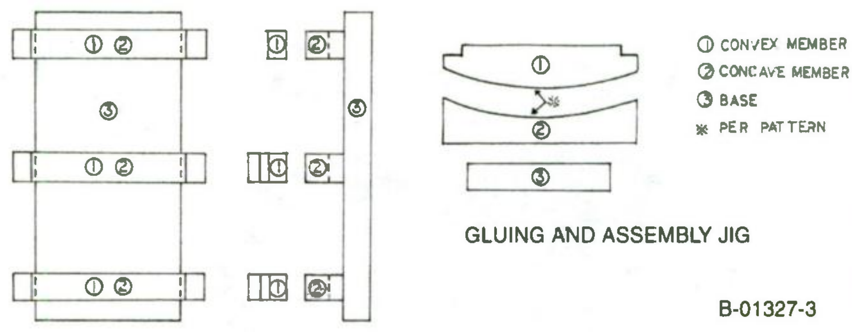


FIGURE 3: Gluing and assembly jig.

PHOTO 5: The gluing and assembly jig.

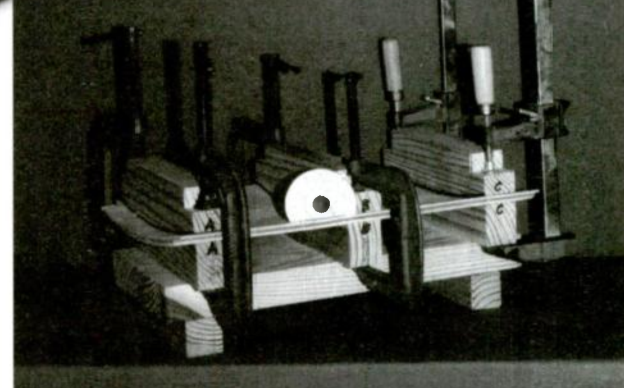


PHOTO 6: The gluing and assembly jig in use.

ASSEMBLING THE HORN

Each side of the horn consists of three or more thin sheets of plywood glued together in the jig. The sides are curved, rigid panels, corresponding to the curvature of the core on which you will assemble them. I recommend forming the sides of at least three thicknesses of ⅛″ plywood, or four layers if they are thinner. Generally, the more layers you use, the less the tendency to "spring back" when you remove them from the jig.

When cutting the pieces of plywood, be sure to cut the material so that the grain of the outside surface of each piece is parallel to the mouth of the horn. Failure to do this may result in breaking the pieces when they are clamped into the jig, or in excessive spring-back when they are removed therefrom. (Experience is such a wonderful teacher!)

Cut the pieces of plywood approximately 2″ longer than the width of the mouth of the horn, thus providing sufficient length to overhang the ends of the adjacent panels when placed on the core. Also, cut the panels approximately 1″ wider than the length from the throat to the mouth of the core, along the side of the horn. You can take these measurements directly from the core assembly.

You will need six C clamps to hold the panels in the jig. Be sure you have these on hand before you start to glue the panels (*Photo 6*).

APPLYING THE GLUE

Assuming your sheets of plywood have a finished side, turn that side of the first piece toward the core. Apply glue to the back of this piece, and then place the second piece over the first. Again, apply glue to the second sheet, and place the third on top of the first two. Whether you are using three or four pieces, be sure to place the finished side of the last sheet *upward*, since this will become the outer face when the horn is completed.

Align the edges of the newly glued panels and secure them in position with a small wire nail driven through the center of each end to prevent them from slipping or shifting with respect to each other when you place them in the jig. Center the glued panels on the jig, and place the convex sections of the jig on top, immediately above and aligned with the concave sections below.

Attach a C clamp to each end of the center set of formers, and tighten slightly. Then apply clamps to the ends of the other pairs of formers, tightening them (and the first pair) incrementally and firmly to prevent buckling of the panels.

If you have only one jig, this would be a good time to take a long break to go fishing, golfing, or do something else. Don't try to rush the process! Let the glue dry thoroughly before removing the panels from the jig. I recommend 24 hours for each panel. Repeat this whole process for each of the three remaining panels.

After the first panel has dried, remove it from the jig. Turn the core assembly upright (mouth downward) and place a ½″ spacer beneath its mouth to allow ½″ overhang of the panels at the mouth. Trim off this excess after you assemble the horn.

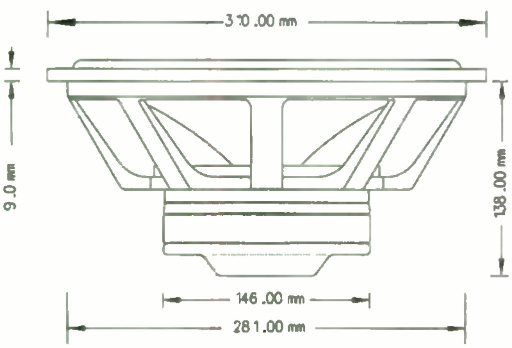
ATTACHING PANELS TO THE CORE

There will always be a slight amount of "spring-back" when you remove the panels from the jig. When attaching the panels to the core, I used three wood screws, #8 × 1½″ long, to secure them. I

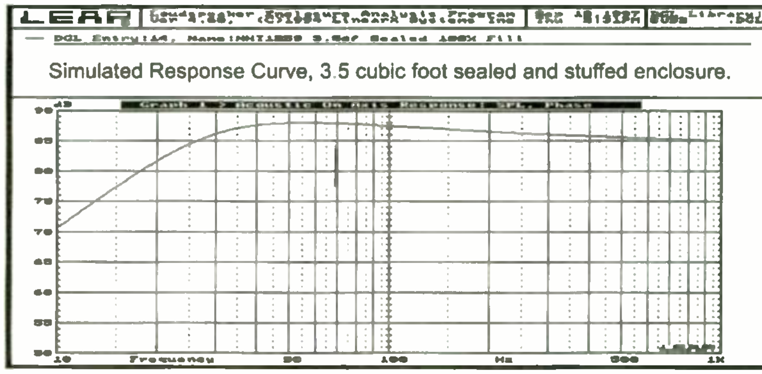
NHT 1259

High Performance 12" Woofer

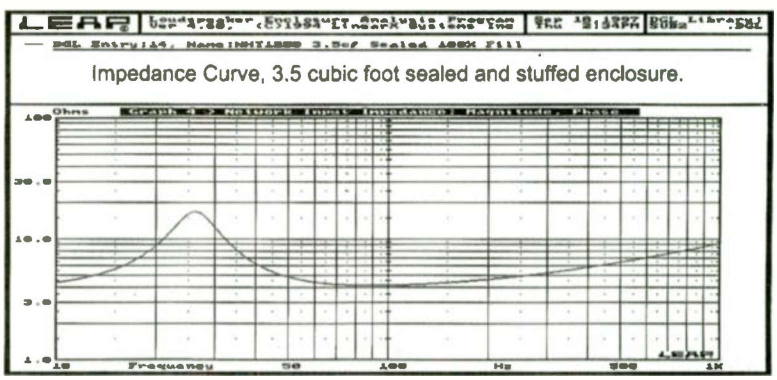
Madisound Speakers is pleased to offer the Now Hear This custom made woofer. The unique characteristics of the NHT 1259 allow it to be used in relatively small sealed enclosures, producing deep and accurate bass to 25Hz. The large voice coil and long excursion insure that this woofer will provide superior transient response with exceptional power handling. The bumped backplate and raised spider prevent bottoming at maximum excursions; the heavy cast frame minimizes energy transfer to the enclosure; and the polypropylene cone with rubber surround promise long term durability in any environment. This woofer is an exceptional choice for any high-end home or autosound system and can also handle the most demanding A/V system.



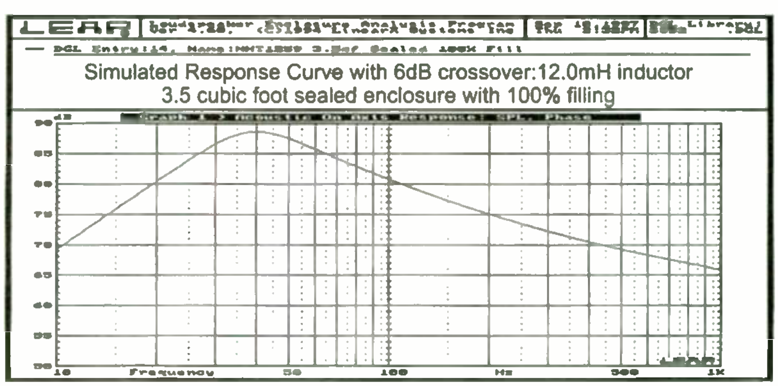
NHT 1259 Specifications	
Fs	16.5Hz
Nominal Impedance	4 ohm
Mmd	128.0 Grams
Cms	696.48 m/n
Vas	238.4 Liters
Rscc	3.52 Ω
Leap Krm	3.277 m Ω
Leap Kxm	10.063 mH
Leap Erm	0.772
Leap Exm	0.743
vcL	1.06mH @ 1K
Bl	9.574 Tm
Qms	2.680
Qes	0.533
Qts	0.445
Voice Coil Height	34 mm
Air Gap Height	8 mm
Xmax	13.0 mm Peak
SD	0.0491 m^3
Surround	Rubber
Cone Material	Polypropylene
Magnet	59 oz.
Voice Coil	50 mm
Music Power	300 Watts
Sensitivity	90 dB 2.83V/1m
Price	**$150.00**



Simulated Response Curve, 3.5 cubic foot sealed and stuffed enclosure.



Impedance Curve, 3.5 cubic foot sealed and stuffed enclosure.



Simulated Response Curve with 6dB crossover:12.0mH inductor
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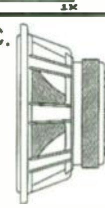


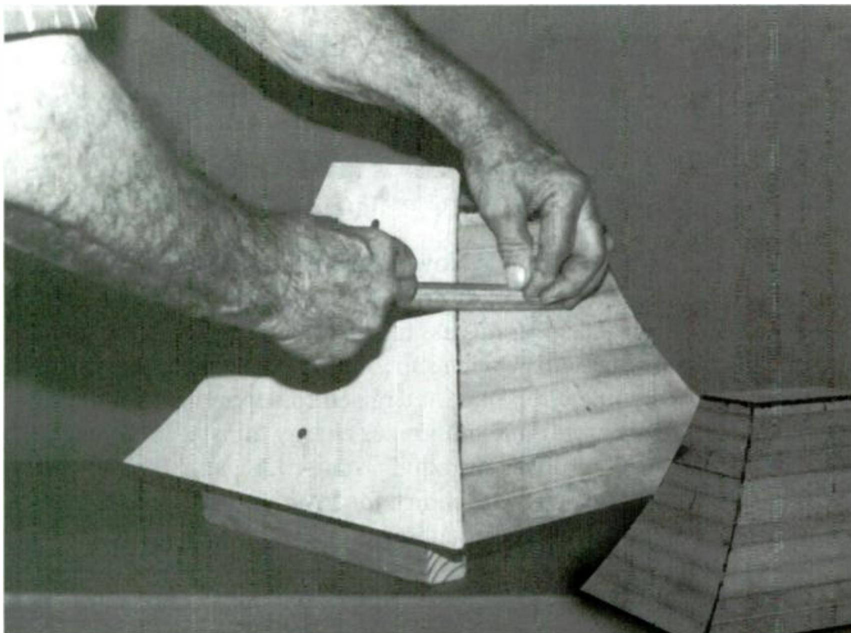


PHOTO 7: Using a fine-tooth rasp to trim the ends of a side panel mounted on the core assembly.

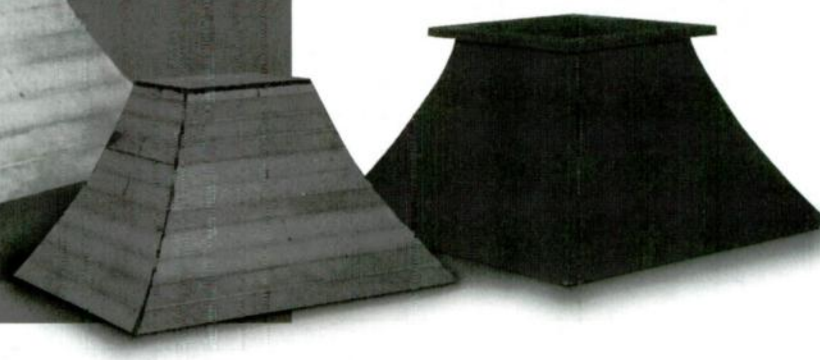


PHOTO 8: The completed core (left) and the horn produced using the core.

put the first screw in the center of the panel, midway from the throat to the mouth, and centered end-to-end. I screwed this in snugly to pull the panel firmly against the core. Then I placed another screw above the first, toward the throat, and a third in line below these two, toward the mouth. These screws hold the panels firmly in place for trimming and fitting.

After installing the first panel and securing it in place, I used a utility saw to trim the excess from each end. Then I used a fine-tooth wood rasp to trim the ends flush with the adjacent sides of the core (*Photo 7*). You must trim the ends flush with the adjacent sides of the core assembly to ensure proper fitting at the corners of the horn. Take your time when fitting these panels–remember that haste makes waste!

After the next panel has been formed and dried, install it on the opposite side of the core from the first panel, attach it as before, and trim and file off the excess in the same way.

Install the two remaining panels in the same manner, with the first two still in place. Trim the excess, and file flush with the outer surfaces of the two original panels. After the trimming is completed, the corners should fit snugly with no gaps between the panels.

FINAL STEPS

Remove the last two panels that you installed. Apply glue to the edges of the two original panels, and then reinstall the last two. Be careful not to allow glue to get onto the core! A strip of masking tape applied to the core is useful to prevent this.

After gluing the panels in place, I used wire nails to fasten their corners together. Be careful to avoid placing the nails where they might interfere with trimming the excess from the mouth or throat later on.

Allow the horn to dry for 24 hours before removing it from the core assembly (*Photo 8*). Before removal, mark around the inside of both the throat and the mouth of the horn where the core and horn meet. Also place a ½″ spacer along the outside of the mouth, and make a mark around the mouth as a guide for trimming the edges of the horn mouth.

Now remove the screws from the panels, and lift the completed horn from the core. Trim the throat and the mouth as required.

To mount the horn on the rear chamber, I cut out an 8″ × 8″ × ½″ collar, traced the pattern of the outside of the throat on the collar, and cut a hole in its center to fit over the outside of the throat. Then position the collar flush with the outside of the throat and glue it in place.

This completes the construction of the exponential midrange horn. Finish it as you please by sanding and applying stain and varnish to the outside. You can fill the holes in the sides with paintable putty and sand them smooth.

You can now admire your handiwork. I think you'll agree that it was an interesting project, and if you are happy with the sound and wish to use it in a stereo system, you can start all over and build another! However, most of the work has already been done: you have already made the calculations, the side-elevation drawing, the core, and the gluing and assembly jig. Have fun!

In Part 2, I will describe another horn built according to the conventional formula, with a smaller throat area.

The second part of this series on transmission lines takes a close look at driver response and the factors that affect performance in a TL system.

Part 2

Transmission Lines: The Real Story

By A. Monk

The TL's system (1m) response is complex both in magnitude and in phase. *Figure 13* shows the slope, peaks, and nulls that define the system, and each wiggle has a meaning. Change the line length or the fiber in the line, and the signature changes. The quantity of interacting variables is the reason it is so difficult to define a mathematical model for the TL. To cut this Gordian knot requires a new view of the TL, not as a linear extension of the response of a piston driver, but of decomposition.

DECOMPOSITION

By decomposition, I mean the view that the TL's system response is composed of two separate signals: the near-field woofer response and the TL's terminus near-field response. This in itself is not very revolutionary. However, if you assume the two signals can be viewed in isolation, you have cut the knot of complexity. The signals must be defined as vectors—phase and magnitude components—and the TL's system response as a vector summation.

I shall next attempt to validate the assumption of decomposition. As a first step, consider how a dynamic driver acts in a TL line. If for the moment you ignore the driver's phase component and look at the magnitude-versus-frequency response, you are back in familiar territory. Driver response is defined by the T/S parameters and constrained by the enclosure volume, resulting in an alignment. The same holds for the TL, and this is shown in *Fig. 14* as computed responses overlaid with measured data.

The graph is a small-signal simulation of a driver for a closed box $Q_{TC} = 0.707$ alignment versus a critically damped, $Q_{TC} = 0.50$ alignment. The dashed line is the response of a KEF B110 driver in a 0.914m line.

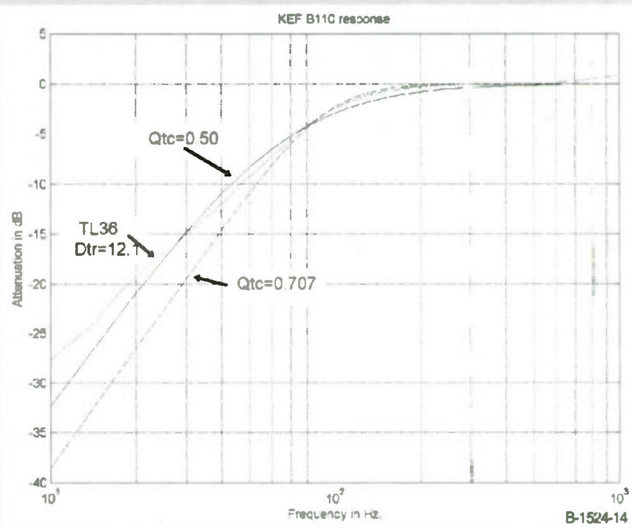
Note that the measured data and the $Q_{TC} = 0.50$ alignment data have very similar responses; they differ only in the low end of the attenuation slope. This is an indication of the large-signal nonlinearity problem[3] that has profound implications for modeling the TL as a linear-phase piston-driver model, but for the present discussion, it can be swept under the rug.

Figure 14 shows that the TL woofer's response is equivalent to a critically damped system, and that you can use a PC driver-simulation program to predict

FIGURE 13: TL 0.914m system response; D_{tr} = 8; Dacron HoloFill II fiber.

B-1524-13

FIGURE 14: TL near-field versus closed-box; Q_{TC} = 0.50.



B-1524-14

the applicability of a specific driver for a TL design.

So you can use the data of *Fig. 14* to do a first cut at a TL design using the KEF B110 driver. At 64Hz, the attenuation slope is ~ −10dB, and assuming that you can realize a terminus gain of about 10dB, you can compute the line length of 1.7m by using the TL's F_R equation. At 90Hz, the necessary gain for a 0dB system response would be 5dB, and the line length 1.2m.

Referencing the impedance data, it is evident that the F_R was shifted lower in frequency, so you can adjust these line lengths to approximately 1.4m for F_R = 50Hz, and 0.96m for F_R = 70Hz.

This last figure is quite close to the measurement for a TL = 0.914m line. Please note that I made a lot of assumptions in deriving this result: fiber characteristics at the F_R frequency, phase linearity for the driver, and TL line-terminus response for the line length. Any of these can drastically change the result.

PURE NUMEROLOGY

As a contrast to the above derivation, consider the following equation that supposedly relates the driver's cone area and T/S parameter Q_{TS} to the TL's parameter of cross-sectional area to define the stuffing density. In his TL design book,[4] Larry Sharp gives a formula that relates driver Q_{TS} to the line-stuffing density:

$$D_S = \frac{(A_{tl} \times Q_{ts})^{0.5}}{S_D}$$

No proof is offered for its validity, and if you assign variable values to the quantities, the results are logically incompatible. This is pure numerology, with no relation to acoustic physics.

I have shown the empirical derivation of driver-type computability with a line length. First you compute the driver's Q_{tc} = 0.5 response from its T/S parameters. Then, noting the frequency where the attenuation slope crosses the −10dB mark, you define the F_R limit. The terminus vector sum, with the driver's magnitude/phase at the −10/−8dB mark, will result in a 0dB relative level when the fiber mass is optimized. At F_R, response is much more complex than the rather simplistic formula in Q&E TL Design suggests.

The important point to realize from this discussion is that the TL woofer's response is equivalent to a Q_{TC} = 0.5 alignment, i.e., a critically damped system. This is why the TL has good transient-response characteristics, as was shown by Bailey's pulse data. A second point is that to define a mathematical model of a TL, you can use the gain and phase data from a PC simulation for a critically damped system to define the driver's near-field response.

The basics of a TL (as shown in *Fig. 15)* consist of the woofer response, the terminus response, and the combined system response. I have added the interior response to examine the assumption that the back of the woofer's response has some relationship to that of the line response.

I have previously stated that you can understand the TL system response (1m) only as a vector sum of the woofer and terminus responses. Instead of going into the mathematics, I'll use the experimentally measured data at the F_R frequency to illustrate this concept. This will also make it obvious why phase information is crucial in a TL design, and why you must use a measured parameter instead of a computed value.

THE WOOFER NEAR-FIELD RESPONSE

Figure 16 shows the phase and magnitude response for the KEF B110 driver in a 0.9144m line. The phase plot is quite linear from 10–300Hz; however, the slope's starting angle is about +10°, and this value is very much driver-level and magnetic-circuit-design dependent.

Note that for F_R = 65Hz, the phase angle = −92.4°, and the magnitude = −6.4dB. This defines the woofer vector at F_R. Note that the phase data is measured, and to relate it to the PC-simulation data,

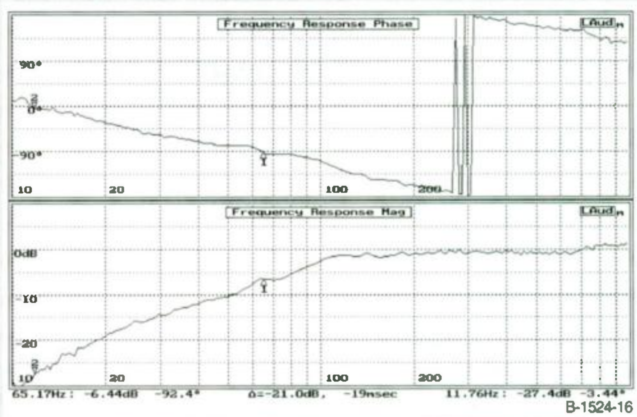


FIGURE 16: The TL near-field woofer response for a line D_{TR} = 8; Dacron HoloFill II fiber.

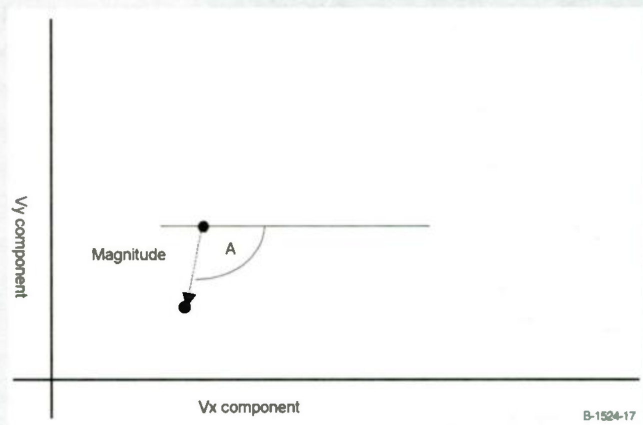


FIGURE 17: Vector representation of woofer near field at F_R.

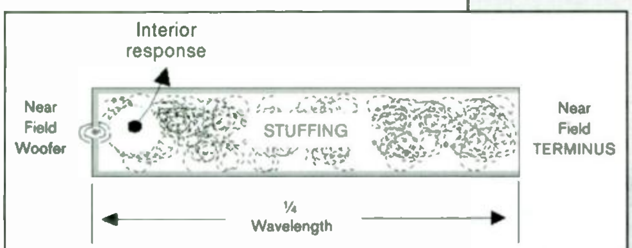


FIGURE 15: The TL model.

TABLE 2

FREQUENCY	D_{TR}	MAGNITUDE	PHASE	
65Hz	8.0	−6.4dB	−92.4°	near optimum
65Hz	12.0	−6.4dB	−83.8°	past optimum
65Hz	16.0	−6.4dB	+95°	way past optimum

you must do a Hilbert transform. *Figure 16* also shows that every point of the woofer's response must be viewed as a complex number composed of magnitude, as shown in the bottom graph, and phase as shown in the top one. This is illustrated in *Fig. 17*.

Table 2 shows the phase angle's dependency on the fiber mass, and *Figure 17* shows the graphic representation of a single point of the data in *Fig. 16* as a vector:

$$V_i = m_i + a_i$$

Thus the magnitude and phase angle can be viewed as a continuum of vector arrows.

Such a continuum forms an array designated by capital letter V (n1:n2), with the numbers in parentheses defining the size of the array. The mathematical manipulation of this representation of the data as a complex number array defines the matrix computation. For a single point, this is graphically illustrated in *Fig. 17*. The addition of the near-field woofer and terminus responses is thus a process as illustrated in *Fig. 18*.

THE TERMINUS RESPONSE

The near-field terminus response in *Fig. 19* is given for a line-stuffing value close to optimum. If you compare *Fig. 19* to *Fig. 7* (Part 1), you see the drastic change that fiber produces in the terminus response. I will discuss later the details of the progressive change in the bandwidth.

Note that for F_R = 65Hz, the phase angle = +13.9°, and the magnitude = +14.9dB. This defines the woofer vector at F_R. To see the dependency of the phase on the fiber mass, look at *Table 3*.

Figure 18 shows how the single vector representing the near-field woofer is added to the vector representing the terminus response to form the vector of the TL's system response.

I hope you can appreciate that the concept of decomposition defines the TL's complexity as a rather simple vector

TABLE 3

FREQUENCY	D_{TR}	MAGNITUDE	PHASE	
65Hz	8.0	+14.9dB	+13.9°	near optimum
65Hz	12.0	+15.5dB	–7.7°	past optimum
65Hz	16.0	+15dB	–162°	way past optimum

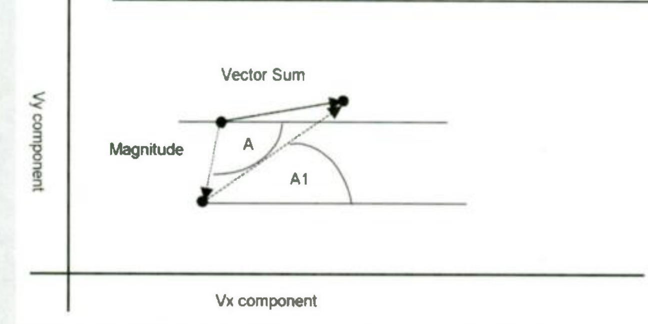


FIGURE 18: Vector representation of woofer and terminus near field at F_R.

TABLE 4

SAMPLE CODE SEGMENT

`V(1:np,3) = A(1:np,5)./sin(A(1:np,12).x(pi/180));`	calculates V2 (scaled phase)
`X(1:np,2) = V(1:np3).xcos(A(1:np,3).x(pi/180));`	calculates the x-component
`V(1:np,4) = V(1:np2) + V(1:np,3);`	V3 = V1 + V2
`X(1:np,3) = X(1:np,1) + X(1:np,2);`	
`V(1:np,5) = sqrt((V(1:np,4).^2) - (X(1:np,3).^2));`	V3 magnitude

TABLE 5

D_{TR}	LF –3DB	HF –3DB	NOTES
1.0	35Hz	120Hz	LF –3dB ignoring the peak
6.7	32Hz	120Hz	
12.0	18Hz	120Hz	

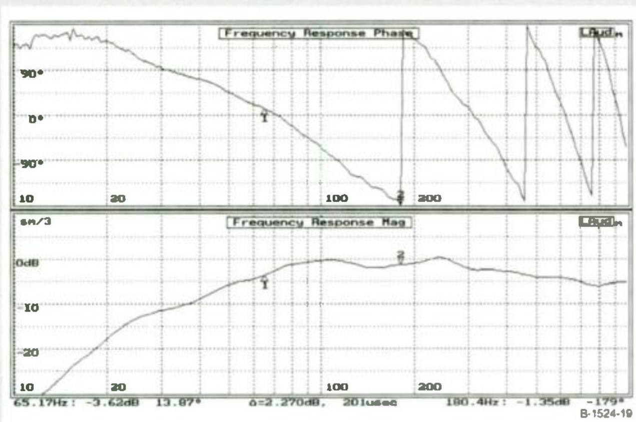


FIGURE 19: The TL near-field terminus response for a line D_{TR} = 8; Dacron HoloFill II fiber.

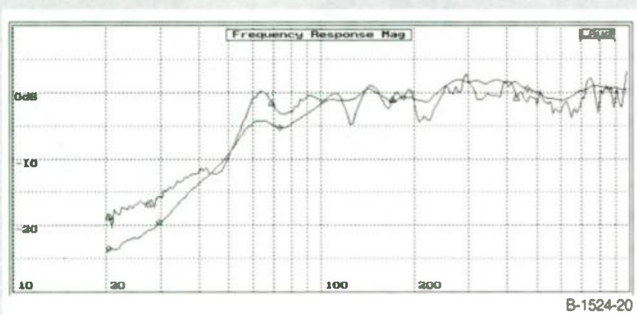


FIGURE 20: F_R gain for D_{TR} > optimum.

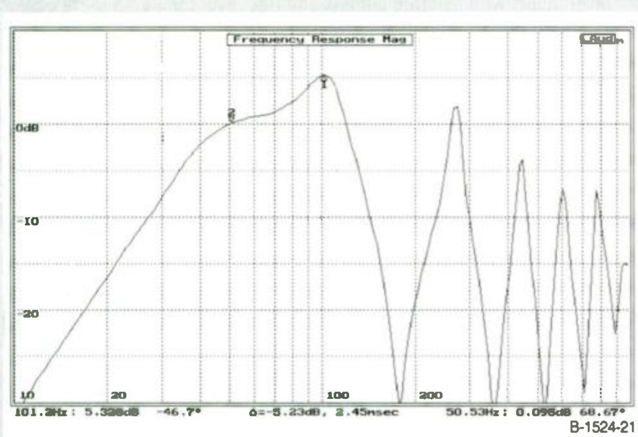


FIGURE 21: Interior; D_{TR} = 1.

summation. At the same time, it defines the fundamentals of a mathematical model.

The mathematics of computing the TL's system response requires a matrix format wherein the vector array is designated by a capital letter and the variables by small letters. Since each vector array can contain about 1000 data points and each TL vector is a product of another vector, the number crunching is formidable and requires a computer. The sample code segment in *Table 4* shows it is a very simple mathematical process. The code describes what the TL does acoustically. The result is shown in *Fig. 13*, the TL system response at 1m.

THE TL SYSTEM RESPONSE

Figure 13 shows a typical TL line response for an optimally stuffed line. The magnitude at F_R is close to 0dB, indicating that the function of gain/phase at F_R of the near-field woofer response is matched by that of the terminus, and that the vector sum can result in a 0dB gain at F_R. Marker #2 is near the null associated with the first harmonic. If the woofer's Q_{TC} = 0.5 response had extra gain, then you could increase the D_{TR} value to minimize the nulls associated with the TL's system response. An alternative is to use a Helmholtz resonant cavity tuned to the first harmonic.

I will briefly touch upon these variations later on in the article. They are very technical implementations that go beyond the basic TL design concepts.

Figure 20 shows the typical effect on system gain when D_{TR} > optimum. As the terminus phase shifts past the optimum angle, the gain at F_R drops. But also note that the harmonic peak/null response decreases, since the attenuation has increased for the higher frequencies. The one exception is the sharp –5dB null at about 130Hz, the shifted phase transition.

In the following section I'll examine the response variations for the terminus versus stuffing density. This then leads to the exploration of the heart of the TL—the fiber characteristics that define the phase changes documented in the previous section.

INTERIOR LINE RESPONSE

Before considering the variations with D_{TR} of the terminus response, I would like to clear up a misconception. Most amateurs would assume as self evident—since the woofers' back response is that of the front *(Fig. 15)* with a 180° phase difference—that the signal at the top of the line would be some facsimile of the near-field woofer response.

The TL throws you a curve, however, showing how dangerous "self-evident" assumptions are. For the unstuffed line, the interior response as shown in *Fig. 21* is that of a resonant cavity with clearly defined harmonic structure. As the line-stuffing density is increased (*Fig. 22*), the line-length harmonics are attenuated, and two characteristics of the interior response are defined: the high-frequency slope knee is a constant defined by ¼λ, and the LF −3dB slope point shifts lower in frequency. *Figure 23* shows the trend for D_{TR} = 12. Also see *Table 5*.

From the data of *Figs. 21–23*, it is obvious that the interior of the TL line's response is not a facsimile of the woofer, but a resonant phenomenon, and that the woofer acts only as an energy source to excite the line resonance. In short, the interior response tends toward a pressure-type phenomenon.

TERMINUS LINE RESPONSE

The terminus response (*Figs. 24–27*) is considered from the unstuffed line up to and slightly past the optimum. The response is that of a bandwidth signal, the low end of which is defined by the line

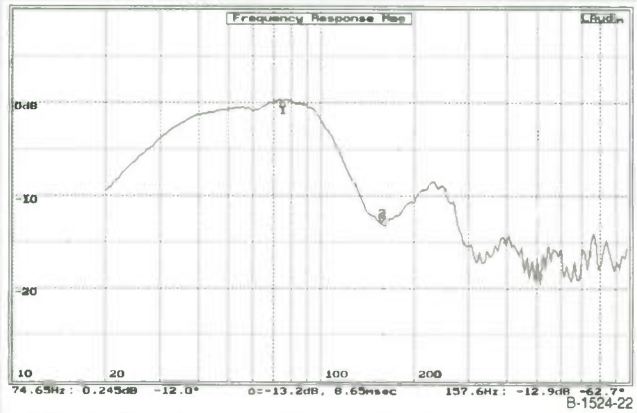


FIGURE 22: Interior; D_{TR} = 6.7.

FIGURE 23: Interior; D_{TR} = 12.

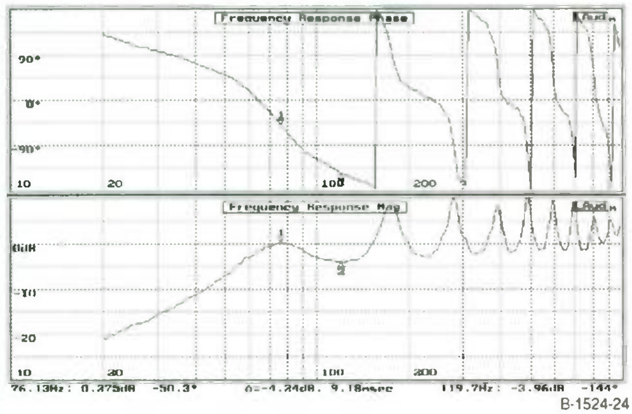


FIGURE 24: Interior; D_{TR} = 1.

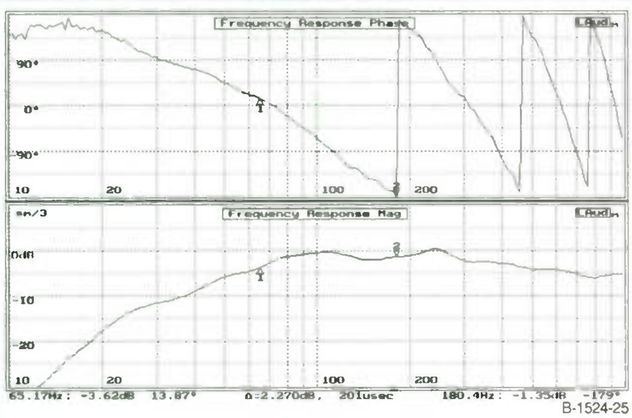


FIGURE 25: Interior; D_{TR} = 6.7.

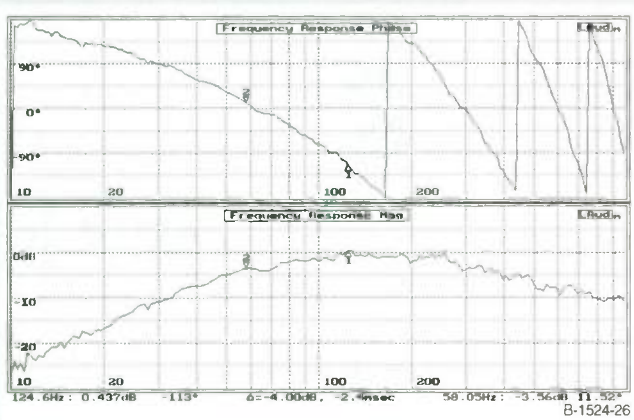


FIGURE 26: Interior; D_{TR} = 12.

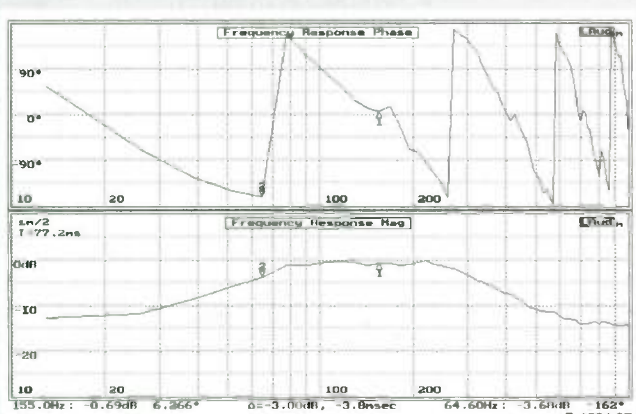


FIGURE 27: Interior; D_{TR} = 16.

length, while the high end approaches an approximately three-octave-wide −3dB point.

Note that the phase is quite linear in the region of D_{TR} >1 to slightly above optimum for the specific TL line length, as is the gain bandwidth curve. As the fiber density increases past the optimum, as shown in *Fig. 27*, the phase exhibits a nonlinearity in phase change, a cusp-type discontinuity.

You should by now have a good understanding of the signal characteristics of the TL and the relationship of line length to the ¼λ resonant frequency F_R. This should allow you to consider the question of the ½-lambda and ¾-lambda line as presented in Table 4.2 of the *Loudspeaker Design Cookbook*.[5]

Suppose you have a friend who comes along and states: "I have read that the ¾λ system is better than the ¼λ design. Bobby, who is an expert in TL design, says that the ¾λ is more efficient and that I can make the line seem shorter just by stuffing it more. He has read that the 'hybrid TL' does this all the time. Bobby helped me design my system for 10.6′ (3.3m) for an 80Hz response, but my RS meter shows the response down by about 10dB at 80Hz. I was hoping you could look at it with your fancy MLS measurements and tell me what's wrong."

He hauls a 10.6′-long TL into my living room for measurements. I suggest that we should try to define the resonant frequency of the line by measuring the unstuffed line. On opening the TL, I find the line has blocks of egg-crate foam for stuffing. He explains that since anechoic rooms use the stuff on their walls, he thought it would be good for killing the high frequencies in his TL. Paraphrasing *Alice In Wonderland*, things are getting stranger and stranger.

On measuring the unstuffed line, we get a slope knee at about 26Hz, and harmonics at 52Hz, 78Hz, 104Hz, 130Hz, and so on. I show him the results and explain that the 3.3m line has a resonance frequency, F_R = 26Hz, and that the second-harmonic peak is at about 80Hz, his supposed tuning frequency for the ¾λ line. Thus the 10.6′ line is acting as a ¼λ, not a ¾λ line. My former friend becomes very angry, saying that Bobby the TL expert designed his TL as a ¾λ, and therefore my measurement must be wrong.

Well, the moral of this story is that you should not believe everything that you read, and the only expert you should believe is Mother Nature. The analogy of the musical instruments (in Part 1) showed that the TL line has a natural resonance frequency that is ¼λ, and though you may mathematically calculate ½- or ¾-wavelength line lengths, the acoustic physics do not change; the TL line has a resonant frequency defined by ¼λ.

TRANSIENT-RESPONSE CHARACTERISTICS

The transient response of a speaker is quite difficult to define. When you hear it, you know it, but it is difficult to measure, since no single phenomenon describes it fully. It is usually defined as rise and fall time for a square pulse, but since speakers are imperfect transducers, they have difficulty reproducing a pulse.

Figure 28 is the step response for a time-adjusted TL. The rise time is that of

FIGURE 28: TL step response.

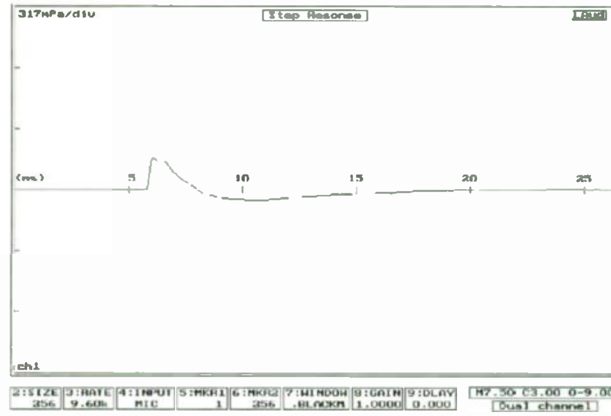


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FIGURE 29: Studio-monitor step response.

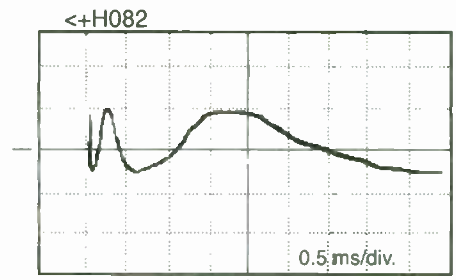

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TABLE 6

PEAK FREQ.	D_{TR}	GAIN AT F_R	PHASE	LF –3DB	HF –3DB	PHASE TRANS
76Hz	1	0dB	–50°	NA	NA	150Hz
96Hz	6.7	–3dB	+15°	0Hz	150Hz	170Hz
124Hz	12	–4dB	+11°	58Hz	300Hz	170Hz
155Hz	16	–4dB	–162°	64Hz	300Hz	64Hz

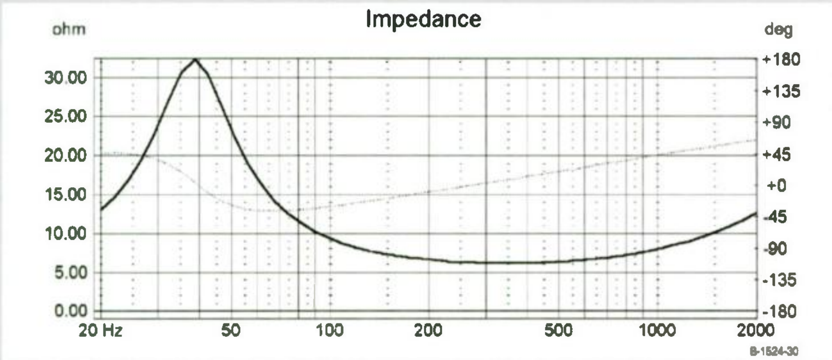

FIGURE 30: Closed-box impedance; Q_{TC} = 0.5.

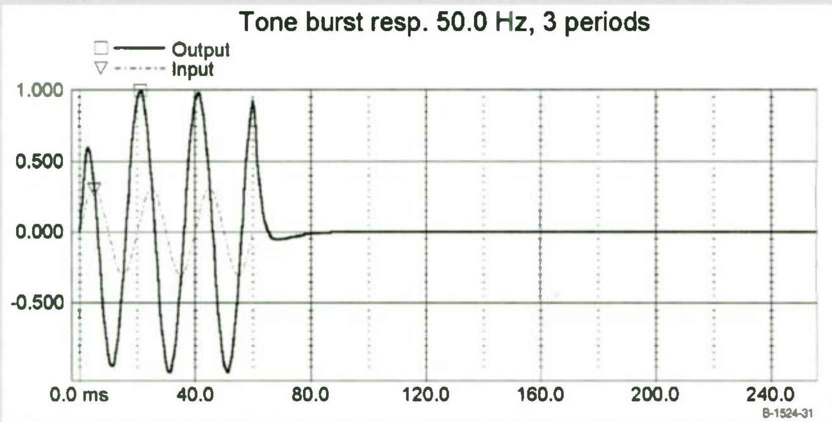

FIGURE 31: Closed-box transient response; Q_{TC} = 0.5.

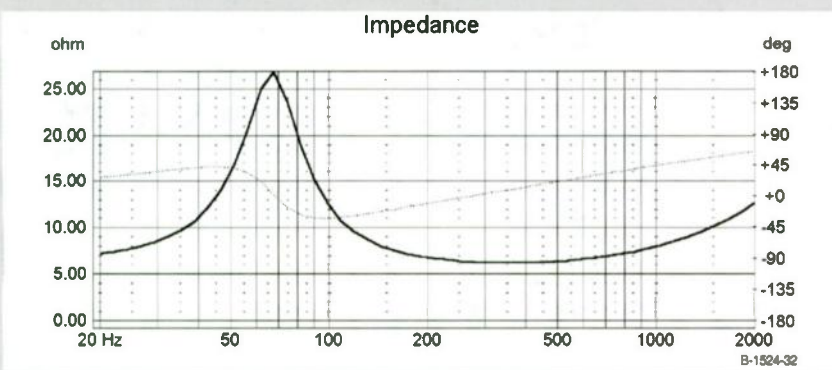

FIGURE 32: Closed-box impedance; Q_{TC} = 0.707.

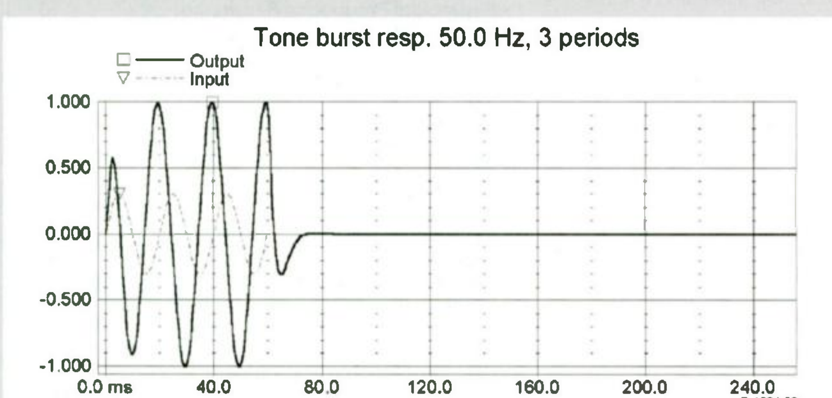

FIGURE 33: Closed-box transient response; Q_{TC} = 0.707.

the tweeter followed by the response of the woofer. A more typical step response (*Fig. 29*) is that of a studio-monitor speaker for a step signal. The drivers are not time-adjusted, so you see three distinct pulses. However, it is significant that each component has a large negative fall-time pulse component, which is an indication of pulse-decay ringing, i.e., that the system is not critically damped. In *Fig. 22*, the woofer's negative pulse extent is truncated, so you can't tell what the fall-time pulse ringing would be, but the indication is that it is considerable.

You can simulate the low-frequency transient response of the system by considering the response of a gated sine for a specific alignment: vented versus a Q_{TC} = 0.707 sealed system. For the TL, the woofer's response loading is approximated by a Q_{TC} = 0.5, the transient response for a critically damped closed box. Such response data is presented in *Figs. 30–35*. Pay particular attention to the response at the point where the impulse (dashed line) stops.

Note that in these figures there is a 90° phase shift between the input (dashed line) and the output at the falling edge. The critically damped response will have some overshoot for ¼ of a cycle, then very rapidly decay to zero. In the step response of *Fig. 28*, this is represented by the negative tail.

Also note that the falling edge has a 140% negative overshoot and requires over 1½ cycles to settle. This data provides some understanding of why the very linear impedance and the minimizing of the impedance peak of the TL data is important.

This concludes Part 2. Using experimental data, I have established the validity of the decomposition model, and also have defined the woofer's response and that of the TL terminus. You have now gained an insight into the fiber-stuffing density effects, providing you with a good foundation for examining what fiber mass does in the TL line. Part 3 will examine the fiber characteristics that will define the optimum stuffing density for the TL.

REFRENENCES

3. Wolfgang Klippel, "The Nonlinear Large Signal Transfer Characteristics of the Electrodynamical Loudspeaker at Low Frequencies," AES preprint 3049 (H-7).

4. Larry D. Sharp, *Quick and Easy Transmission Line Speaker Design*, Mahogany Sound, 1993.

5. Vance Dickason, *Loudspeaker Design Cookbook*, Audio Amateur Press, Old Colony Sound Lab, P.O. Box 876, Peterborough, NH 03458, 1995, p. 75.

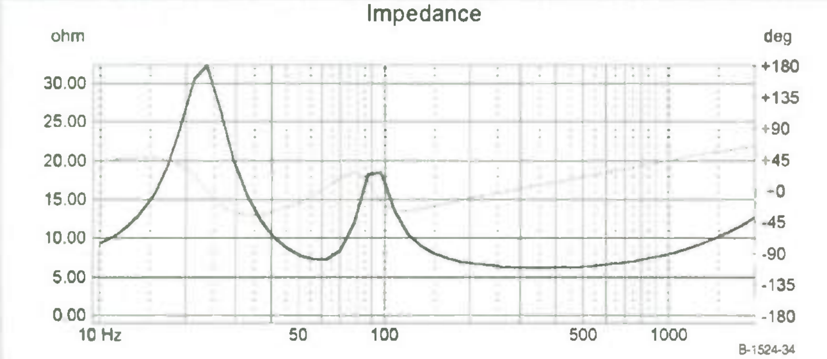

FIGURE 34: Vented-box impedance.

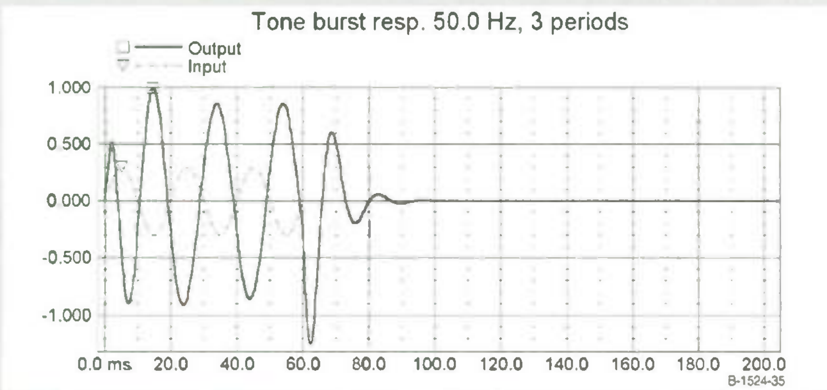

FIGURE 35: Vented-box transient response.

Here's a postscript to an article we published a decade ago. This Danish designer presents some groundbreaking work in the area of distortion-canceling feedback in small systems.

Acceleration Feedback Systems

By Hans J. Klarskov Mortensen

Those who take an interest in this rather remote corner of audio design may remember my article in *SB* 1/90[1] in which I describe a DIY way of experimenting with acceleration feedback in relatively small bass systems. Since the publication of that article, I have discussed the idea with quite a few people, but I do not claim that servo control of woofers is the "best" in any meaning or aspect of the word.

As I outlined in the article, the commercial success of the idea has been limited—to put it mildly. However, there are very ambitious systems around, e.g., Entech's massive VSW-1 system, and it seems that the idea simply won't die. One reason for this could be that it is an interesting technical challenge with a certain promise.

All sorts of objections to it could be raised, but I do not intend to discuss them here. If you are interested, you can read along and experiment as you please. Sometimes the desirability of belonging to a club is inversely proportional to the size of that club's membership—just to rephrase a film star's words in technical lingo.

THE FEEDBACK ADVANTAGE

The main advantage of feedback in loudspeaker design is the possibility of making a relatively small speaker play low frequencies at reasonable sound levels with reasonably low distortion. The costs include added circuit complexity and a need for potent power amps with lots of power and voltage swing. Since feedback in loudspeakers is worthwhile only at low frequencies, most feedback systems make do with two power amplifiers: one for low frequencies and the other for the middle and high ranges.

Of course, this increases both complexity and cost—though the need for elaborate passive crossover networks is eliminated. In many contexts, these problems are very minor, and they are easily outweighed by the improved quality of the low frequencies. There are, however, situations when it would be nice to make do with just one amplifier—for example, in small PC speaker systems or small compact monitor systems. Another possible customer could be the serious audiophile who likes neither active crossover networks in the very sensitive lower-midrange frequencies nor several different-sounding amplifiers.

After I published the article in *SB* 1/90—and a somewhat revised version in the Danish magazine *High Fidelity*—I was contacted by a small company that wished to develop a very compact monitor system, specifically one with only a single power amplifier, but with the possibility of manipulating the line-level signal. Could I do that, please? I promised to try. This is what I came up with.

SIMPLE SINGLE-AMPLIFIER SERVO SYSTEM

My first thought went along the lines suggested by D. De Greef and J. Vandewege in their article "Acceleration Feedback Loudspeaker"[2] (the schematic

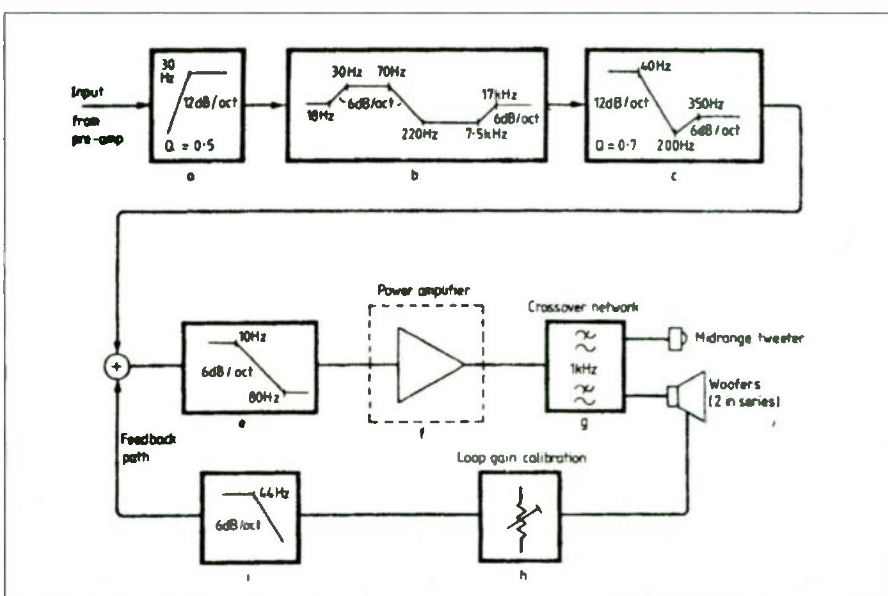

FIGURE 1: Block diagram of De Greef and J. Vandewege's acceleration feedback system. (Reprinted with permission from *Electronics and Wireless World*.)

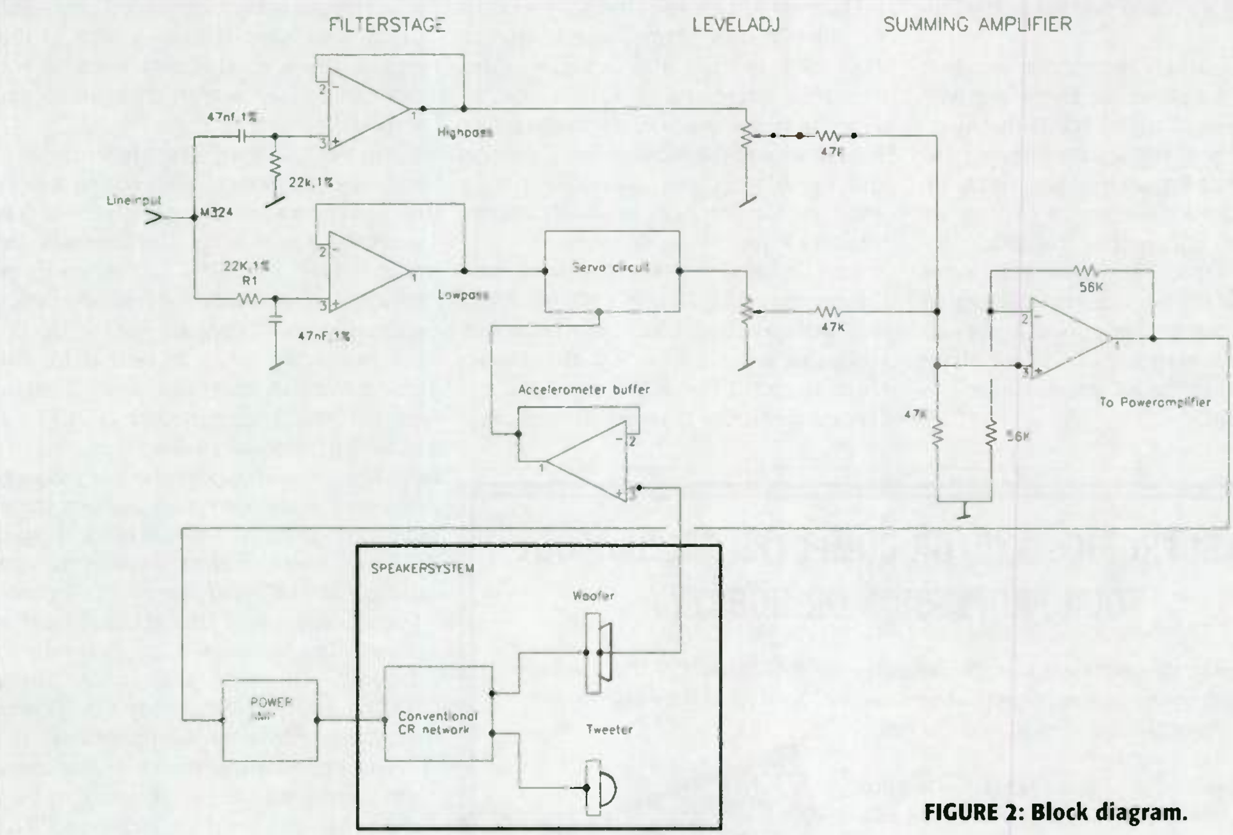

FIGURE 2: Block diagram.

of their project was reprinted in my article in *SB*).

De Greef and Vandewege use just one power amplifier for the whole system. The serious drawback is that it requires a very complicated filtering circuit that must be tailored absolutely individually to each drive unit with its accelerometer. *Figure 1* is the block diagram of De Greef and Vandewege's system. Despite the very careful tailoring of the compensation networks, it goes almost without saying that a lot of phase shift and other things are more or less out of control.

However, the feedback system is worthwhile only below, say, 100Hz, so the trick is to exploit its benefits in that frequency range and prevent it from interfering with the rest of the audio range. This is to some extent what De Greef and Vandewege did, but since I am a great one for simplicity in audio design, I went for a much simpler solution.

I retain filtering as the first stage after the preamp. The filter is a simple 2-way 6dB/octave electronic crossover network that separates the low frequencies from the mid-to-high frequencies. I chose a first-order network here because it is steep enough, provided the −3dB point is sufficiently low. A filter of this type is also a good choice because you can easily make it with great phase and response accuracy.

After this simple filter stage, the mid and high ranges are directed to a summing stage and then passed on to a conventional power amp. The low-pass section is connected to the servo loop, which is responsible for both distortion reduction and increased linearity in the low-frequency range, as well as for flattening out the peaks that result from an excessively high speaker Q, as I explained in my original article.

After the servo loop, the low-frequency signal is fed into the summing stage, the level controls of which make it possible to achieve perfect linearity around the initial first-order filter's −3dB point. The advantage of this arrangement is simply that feedback is applied only to the low-frequency range below about 100Hz, and you need only one power amplifier to drive the whole system.

The crossover network in the more complete loudspeaker system can be a conventional passive network. The crossover frequencies of this network have no bearing on the purely electronic

FURTHER SOURCES

In the interest of science I have reprinted all the references from the original article, plus some new literature I have found. The list is still not complete, though.

1. Siegfried Linkwitz, "Loudspeaker System Design," *Electronics and Wireless World*, May and June 1978.

2. Jean Hiraga, "Le Preamplificateur SRPP," *Selection de L'Audiophile*, Tome 1: L'Electronique, Paris, 1985.

3. R. Conell, "Feedback in Loudspeakers," *Elektor Electronics*, April 1987.

4. Elbert Hendricks, "Et EMK-MFB Bashøjttaler System," *High Fidelity*, No. 11 and 12, 1979, Copenhagen, 1979.

5. J.A. Klaassen and S. H. de Koning, "Motional Feedback With Loudspeakers," *Philips Technical Review*, Volume 29, 1968, No. 5

6. Arthur Brown, "Servo-controlling the AR1," *SB* 3/89.

7. H.W. Holdaway, "Design of Velocity Transducer Systems for Stable Low-Frequency Behavior," *IEEE Transactions on Audio*, September–October, 1963.

8. W. Holle, "Gegenkopplung an Lautsprechern," *Funk-Technik*, Nr. 18, 1952.

9. Richard E. Werner, "Loudspeakers and Negative Impedances," *IRE Transactions on Audio*, July–August, 1958.

10. Robert L. Tanner, "Improving Loudspeaker Response with Motional Feedback," *Electronics*, March 1951.

11. Hans J. Klarskov Mortensen, "Basgengivelse—pro et contra 3. Del," *High Fidelity*, March 1993, København.

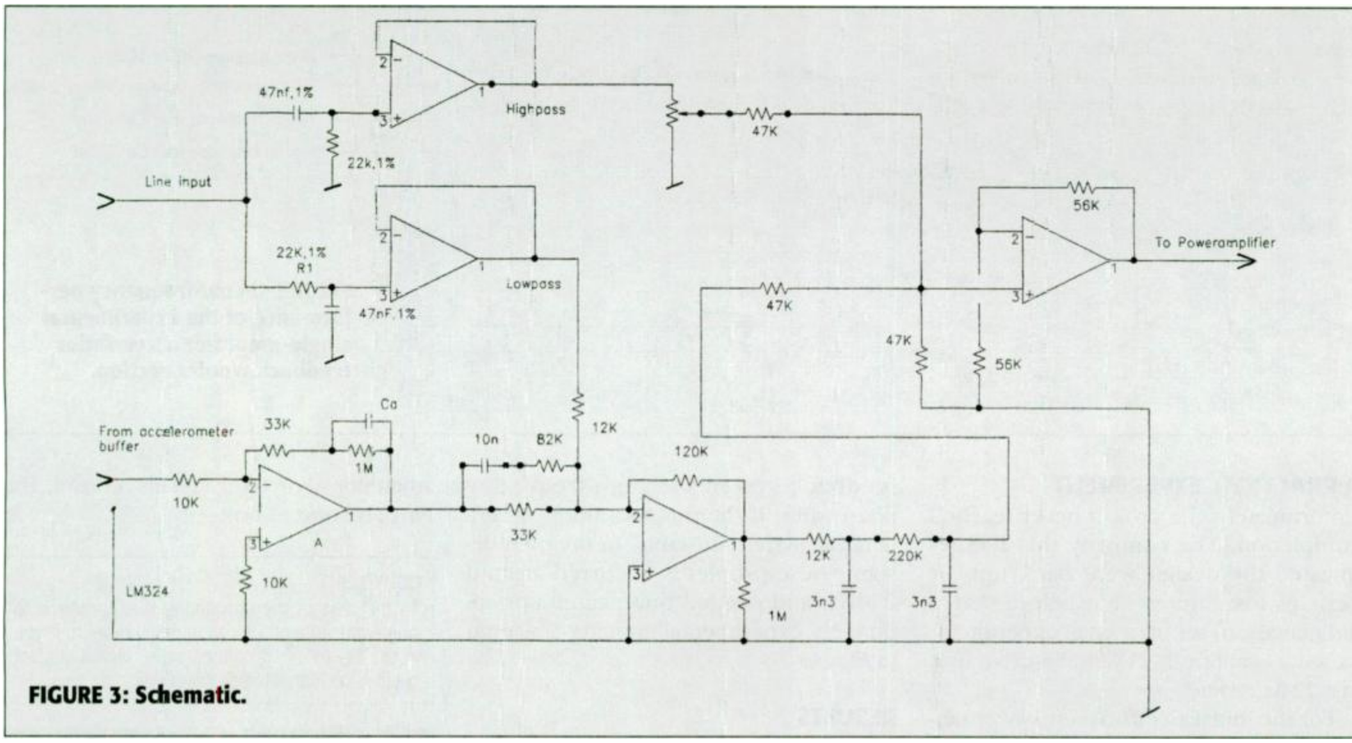

FIGURE 3: Schematic.

manipulations of the line signal. Thus, insofar as the circuitry described here is linear with respect to the speaker system's actual woofer, the quality of the speaker is all that matters. You might say that the integrity of the speaker is unharmed—apart from the very slight change of woofer parameter caused by the tiny accelerometer. But since this involves the addition of only a few grams of extra mass to the woofer's cone, in most cases you can neglect it.

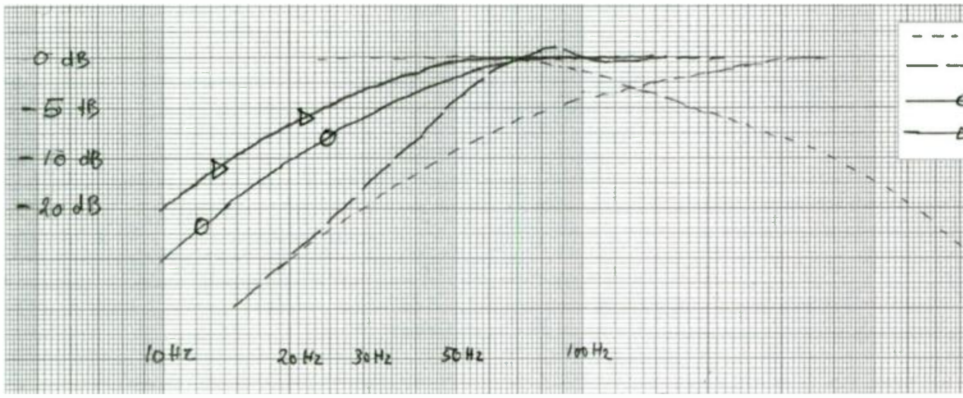

FIGURE 4: Low-frequency performance of the experimental single-amplifier acceleration feedback woofer section.

A PRACTICAL EXPERIMENT

Unfortunately, the project never reached completion. The company that had requested the design went bankrupt, or perhaps lost interest, but before that, I did manage to set up a small experiment. I used a ScanSpeak 21W/8554 drive unit in a 22-ltr cabinet.

For the initial electronic crossover network, I chose a −3dB point at around 130Hz. This means that the range in which the servo system is active is very small indeed, thus making the filtering and control of the inevitable phase shift much easier to handle. A problem in all feedback systems is high-frequency phase shift: if there is too much, it renders the system unstable. In my own design, these problems occurred around 250Hz, and needed quite careful attention. My experimental circuitry is shown in *Fig. 3*.

RESULTS

Figure 4 shows the low-frequency performance of the experimental setup, measured "by hand" with a Bruël & Kjaer condenser measuring microphone. (This was before computer-based systems became economically possible for amateurs.) For more details, consult the Further Sources box.

REFERENCES

1. Hans J. Klarskov Mortensen, "An Acceleration Feedback System," *SB* 1/90.

2. D. De Greef and J. Vandewege, "Acceleration Feedback Loudspeaker," *Electronics and Wireless World*, September 1981.

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West Europe Lowther Club
Victor Meurisse
Avenue Plissart, 16
Brussels 1040
(011) 32-2-736-7394

BRAZIL

Audio Club de São Paulo
Av. Ver. José Diniz 3135 cj 92
Campo Belo, São Paulo, SP 04603-020
(011) 530 1472, FAX (011) 530 8465
E-mail techmark@dialdata.com.br

CANADA

Montreal Speaker Builder Club
Andrew McCrae
4701 Jeanne Mance
Montreal, PQ H2V 4J5
(514) 281-7954

Toronto Area DIY Speaker Club
Michael Mansor
(416) 626-9132
E-mail james.lcr@sympatico.ca

CHILE

Christian Bargsted
Presidente Kennedy 3650
Apt. 101, Vitacura, Santiago
(011) 562-207-0673
FAX (011) 562-538-0638
E-mail centroac@chilnet.cl

FRANCE

Sound Reinforcement Club of Paris
Laurent Bonnet
11 rue de la fontaine du but, Paris 75018
Phone/FAX (011) 33-1-42-23-71-47

GERMANY

ESL Builders Group
Gunter Roehricht
Bühlerstr. 21, Böblingen 71034

Profi Club Visaton
Bettina Schaaf
Visaton Lautsprecher
Ohligser Str. 29-31, Haan 42781
(011) 49-2129-522-0
FAX (011) 49-2129-522-10

HOLLAND

Lowther Club Holland
Bert Doppenberg
Brinkersweg 16, 8071 GT Nunspeet
Phone/FAX (011) 31-341-254500

Audio Vereniging Midden Nederland
Hans van Veenhuizen
Ir.P. Calandstraat 12, 3841 KA Harderwijk
(011) 31-341-416599

INDONESIA

Silo Ultac Audio Society
Febri Arianto
PO Box 6567/SMS, Semarang 50065
(011) 62-811-280060
FAX (011) 62-24-7621647

Tube Audio Enthusiasts
Ignatius Chen
Setrasari 7, Bangdung 40152
FAX (011) 62-22-213277, 62-22-210408
E-mail dutar@digicron.com

ISRAEL

Audio Center
c/o Kobi Cohen
11, Hayey Adam St., Tel-Aviv 67499
(011) 972-3-696-5054
E-mail audio_c@inter.net.il

SOUTH AFRICA

Hi-Fi Club (Cape Town)
Chris Clarke
PO Box 18262, Wynberg 7824
(011) 27-21-7618862
E-mail chrisc@iafrica.com
Website http://users.iafrica.com/j/je/jessica/hificlub

TURKEY

Istanbul Hi Fi Club
Adnan Arduman
(011) 90-216-310-44-70
FAX (011) 90-216-343-42-01

UNITED KINGDOM

London Live DIY Hi-Fi Circle
Lance Dow
E-mail dow@ram.co.uk
Website http://www.geocities.com/CapeCanaveral/9170/index.htm

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PO Box 876
Peterborough, NH 03458-0876
FAX (603) 924-9467
E-mail editorial@audioXpress.com

Don't be misled by manufacturers' sales hype when you go shopping for new drivers. Remember: caveat emptor.

Part 2

Navigating Speaker Design:

Sleuthing Driver Parameters

By Mark Wheeler

Amateur loudspeaker builders usually suffer from the limitations of having neither the facilities nor the budget to design and manufacture their own drivers. Budget constraints are also likely to limit the number of driver options you can purchase and test for a given project. Furthermore, rarely do you have any opportunity to listen to sample drivers in any application at all, let alone one bearing any similarity to that which you intend to build. The result is that you must rely heavily on the data provided by manufacturers when selecting drivers.

Manufacturers' data comes in two basic forms: actual numerical data or sales hyperbole. The former varies in quality. At best, some manufacturers provide tightly defined test data with stated conditions and tolerances. At worst, some suppliers provide vague promises of "frequency range" or "music power handling." The latter kind of sales data is completely useless, and if that is all that's available, I wouldn't buy products from such vendors.

DECODING SALES PITCHES

Sales hyperbole may actually be far more useful than it may first appear. The language and content may provide a good indication of the designer's philosophy if the literature originates from the manufacturer (or a local market translation of the manufacturer's own material) rather than from an importer or other intermediary. The original design priorities are likely to be those most heavily emphasized, and if they coincide with those of your project, their drivers might be suitable for your short list. Conversely, philosophies that appear contradictory to those of your planned project would indicate you should avoid such products.

Many designers develop an instinct for manufacturers' sales literature that often proves uncannily accurate. Alongside the usual tables of Thiele/Small signal parameters and thermal specifications, there is usually a brief (or sometimes not so brief) treatise on the reasons why the product is the best of its class by such a wide margin that any customer would be foolish to look elsewhere. Some imagined examples:

"The Plunkk Eight-O die-cast chassis supports a massive magnet. The Plunkk Eight-O also features an edge-wound copper voice coil on a smelt-proof former bonded to the composite fiber cone with adhesives developed by NASA to resist reentry temperatures. It is the best 8″ bass driver in the universe."

"The Silkko 2020 uses the latest damped mineral-loaded coco-polymer super-hysteric cone for total freedom from resonances, developed using laser interferometry. The soopa-gloop damped neoprene inverted roll surround provides freedom from reflected travelling waves. It's the 8″ driver with the cleanest midrange in the universe."

"The Tankk Titanium 20 has an ultralite titanium cone; Superflexx™ surround; six-nines purity copper voice coil; and silver litz lead-out wires. No other 8″ driver in the universe is more musical."

The Plunkk Eight-O probably has high sensitivity, high power handling, high f_s, and powerful dynamics. So if a smooth and even frequency response, low coloration, and low frequency extension are your goals, then look elsewhere—perhaps at the Silkko 2020.

The Silkko 2020 claims to be "clean," which probably means low coloration and neutral frequency response, suitable perhaps for small studio monitoring and orchestral works. But the trade-offs to reduce all forms of unevenness and coloration may have limited its capacity to handle fast dynamic swings.

The Tankk 200 may have excellent resolution of low-level information or "inner detail," to use the bizarrely molested language of hi-fi consumer-magazine reviewers. It might "image" well, constructing precise audio virtual images of instrument positioning. But if you desire to fill a large room with dub reggae, you'd be better off plonking the Plunkk on your short list.

You have probably been doing this intuitively for years. Perhaps you also match the drive units in a system by a similar process.

Optimizing system bass response and loading from published and measured parameters are familiar tasks to *Speaker Builder* readers, but there are many more diverse nuggets of invaluable design information lurking among the small-signal parameters.

RESOLUTION AND DYNAMICS FROM T/S PARAMETERS

Some manufacturers (e.g., Focal) publish a figure for cone-assembly acceleration, symbolized by the uppercase Greek letter Gamma, Γ, with the units $ms^{-2}.A^{-1}$ (ms = meters per second; A = driver acceleration per unit of current applied).

Simply put, this figure indicates the rate at which the drive unit's moving parts can accelerate when a given signal is applied to the voice coil.

In audio terms, this translates as the small-signal resolution of the motor system. The faster the cone accelerates, the closer it can translate the electrical waveform into movement; i.e., the more accurately it can reproduce an analog of the electrical signal. My own experience backs this up, and I confirmed it with a little experiment: I tried two drivers from the same manufacturer having different "acceleration" figures but otherwise similar construction (same chassis, magnet, and cone material) and specifications, working within their intended passbands. The passbands were further defined by active crossovers.

Driver A worked out to be approximately $\Gamma = 650\ ms^{-2}.A^{-1}$, while driver B was approximately $\Gamma = 390\ ms^{-2}.A^{-1}$. Both were described as bass/midrange units, but I used them only as 200Hz–2.4kHz midrange units in order to minimize the effects of their very different bass response and power handling. I auditioned each only in mono, in an active system with level matched in each case. Driver A sounded more detailed and more transparent, but especially more dynamic.

It was as though driver B were subject to some compression. But driver B has less acceleration because it has a lower Bl product as a result of a longer voice coil for sealed-enclosure applications and much higher power handling. Driver A has a shorter voice coil for reflex boxes and a slightly higher sensitivity that trades off against its lower power handling. In the restricted application tested, driver A was consistently superior in listening tests.

The manufacturer makes these two versions of the same drive unit for very different applications, and it is unlikely that you could choose between them for a particular project. I use the example to illustrate the difference in musical qualities between two different acceleration capabilities when all other factors are controlled. With a specific magnet, a smaller-diameter and shorter voice coil will have a higher Bl product and hence higher acceleration, but larger-diameter and longer voice coils are needed for higher power handling, so "you pays your money and you makes your choice."

CONE MATERIAL

Many other factors affect perceived resolution, although to a lesser extent than the motor system. Cone material, for example, has long been associated with this aspect of driver performance because of different stiffness and break-up modes. Cone-material influence is easily identifiable by using similar chassis/motor assemblies with different cones. The elasticity of the cone renders the acceleration figure meaningless at higher frequencies, since the cone ceases to behave as a single piece at the instant current is applied.

Modern stiff materials, such as various reinforced composite fibers including carbon fiber and Kevlar®, do extend the piston-like behaviors of the cone assembly to ever higher frequencies (in contrast to floppy old lightweight paper cones with annular rings) to encourage controlled break-up with increasing frequency.

Within a given set of operating limits, most especially the upper frequency limit required, faster-accelerating (that is, with a higher gamma figure) drivers usually sound more dynamic and musical than their otherwise similar counterparts. Many of the early attempts at heavily coated, low-coloration cones gained notoriety in Britain for sounding slow and undynamic compared with their paper-coned immediate predecessors.

In the relentless pursuit of neutrality above all else, these drivers had indeed been neutered and neutralized. Acceleration is just one of the factors you must balance among many, but for music reproduction it should be high on the list of priorities for a given operating range.

Acceleration is calculated by the formula

$\Gamma = Bl/Mms$ (acceleration = magnetic product divided by moving mass)

Hence, big magnets, short voice coils and gaps, and small, lightweight cones inevitably lead to higher cone-acceleration capabilities. At first glance, they also correlate with higher sensitivities, but there are more factors at work, so a driver with higher sensitivity is not necessarily a faster-accelerating driver.

Γ does not allow comparison of drivers with different areas operating over different bandwidths. It does not give any indication of the effect of driver size. It is possible to have a Γ range from 500 $ms^{-2}.A^{-1}$ to 1,500 $ms^{-2}.A^{-1}$ in 4″ drivers and from 200 $ms^{-2}.A^{-1}$ to 600 $ms^{-2}.A^{-1}$ in 8″ drivers within the same manufacturer's products. How will these integrate with one another? Given their differing upper-frequency limits, how can they be compared?

If motor-acceleration Γ is a useful indicator of small-signal dynamics, one step further might be a useful indication of overall dynamics. Such an indication might arise from a drive unit's ability to accelerate air (while the cone is acting as a piston). Multiplying cone acceleration by cone area provides "air-volume acceleration," Γ.a expressed in $m^3s^{-2}.A^{-1}$:

$$\Gamma.a = Bl.a/Mms$$

Hence, big magnets, short voice coils and small magnet gaps, and large, lightweight cones inevitably lead to higher air-volume acceleration Γ.a capabilities.

ACCELERATING AIR

"Ah! looking at that equation, isn't it just expressing efficiency?" demands cynical reader from stage left.

No it is not; it is about accelerating air, not just moving quantities of air. I have ignored various factors, including any consideration of compliance or friction. Actually, the factors involved are broadly similar, so usually one does tend to follow the other as night follows day. But there are many major exceptions, and two drivers from the same manufacturer may have similar efficiencies, but quite different air-volume acceleration Γ.a.

The correlation between air-volume acceleration and audible dynamics is very strong. I tried diverse drivers from several manufacturers, and they fitted the hypothesis without exception:

- bass/mid speakers that scored Γ.a above 12 $m^3s^{-2}.A^{-1}$ seemed to reproduce full orchestral scale and uncompressed recordings reasonably effortlessly;
- those with Γ.a above 9 $m^3s^{-2}.A^{-1}$ are also reasonably good in both scale and transient impact, particularly with smaller-scale ensembles of all musical genres; and
- those with Γ.a below 7 $m^3s^{-2}.A^{-1}$ sounded dull or compressed.

A few drivers scored Γ.a between 15 $m^3s^{-2}.A^{-1}$ and 20 $m^3s^{-2}.A^{-1}$, and these proved to be outstandingly dynamic, although many suffered coloration because of big, light, paper cones. The Focal Audiom 7K midrange driver with Γ = 1220 $ms^{-2}.A^{-1}$ and Γ.a = 20.1 $m^3s^{-2}.A^{-1}$ managed low coloration with high acceleration-factor Γ and air-volume acceleration Γ.a, and demonstrated that all these goals are worthwhile without compromise—the sound is lively, musical, and very dynamic.

PARAMETERS IN SYSTEM DESIGN

Many system builders play safe with the bass and midrange drivers of three-way systems by choosing them from one range of one manufacturer. There are many good reasons why this is a useful and safe practice. The drivers should integrate well, because the same designers will have used similar design goals and priorities. Most cone materials have their own coloration signatures, so to play safe, these also should be matched.

One other observation that arose during these trials concerns drive-unit matching in systems. I tried only a few combinations because of limited availability. Poorly matched air-volume acceleration between drivers in a system, especially between bass and lower midrange, did not sound homogeneous, and the system sounded ill timed, a bit like an ensemble of good musicians in the early stages of rehearsal of an unfamiliar piece.

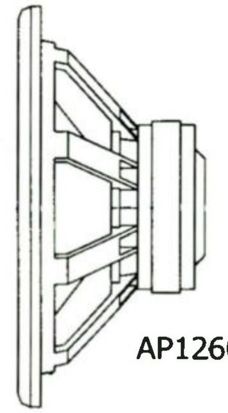
If the driver with the higher Γa was the bass driver, the music seemed to be running away with itself, but if the one with the higher Γa was the midrange, the bass sounded turgid and lagging behind. When the Γa was similar for both drivers, the system sounded more coherent, whether or not it was at the faster end of the range.

Being able to put numbers to a phenomenon, however much they may be ball-park figures rather than precise measurements, will allow amateur system designers to be even more informed at the planning stage of their system building.

All these figures indicate a drive unit's performance under small-signal conditions. As the applied signal increases, other factors begin to affect performance. Large-signal driver effects include: voice-coil length and heating; pole-piece gap height and shape; and cone-excursion limits. Large-signal system effects include bass-loading type and passive crossover-component behavior, in addition to the effects in each driver.

So home-system builders can derive more than just hyperbole and bass alignments from manufacturers' literature. Cone-assembly acceleration Γ gives a useful idea of driver-resolving power within a given passband, whether you call it "inner detail" or "microcontrast." Air-volume acceleration Γa provides amateur designers with one more tool to contribute to system integration and "dynamic shading." These are just two more considerations to ponder when you leaf through the manufacturers' specifications to select the drivers for your next project.

Part 3 examines some aspects of the construction of boxes in which to put your chosen drivers.



Here's another perspective on the speaker-cable controversy. This author shows you how to construct your own with low resistance and low inductance.

Do I Need A New Speaker Cable?

By Jesse W. Knight

I did not set out to make speaker cable. Quite frankly, I believed enough had been said already; however, my current project led me to realize that this was not true. I was mistaken in my belief that 0.3Ω of cable resistance would affect the woofer response only near the lower f_3.

A STICKY MESS

All too often my speakers end up in my woodworking shop, and the sawdust and plaster dust sticks to the midrange, adding mass. I needed a cleanable, inexpensive dome with no sticky treatment.

The Vifa 3″ dome turned out to be a perfect solution. This dome is so different from any driver I have worked with that I decided to try many network designs with it over a long period of time. I designed a variable capacitor and variable inductor network to determine the best way to use it. I was rewarded with better sound than I ever expected, at very low cost. During this time some interesting problems arose.

Variable networks introduce variable resistance, and this becomes a huge problem. Some network-driver combinations are extremely sensitive to resistance. Changes of 0.2Ω produce audible changes that are easy to measure.

Ironically, the most sensitive I encountered was a conventional 350Hz third-order Linkwitz-Riley low-pass with a Pyle 12″ woofer. Even worse, added resistance produced a 100Hz peak, not the expected small change at the lower f_3. Low-resistance inductors and cable became a high priority.

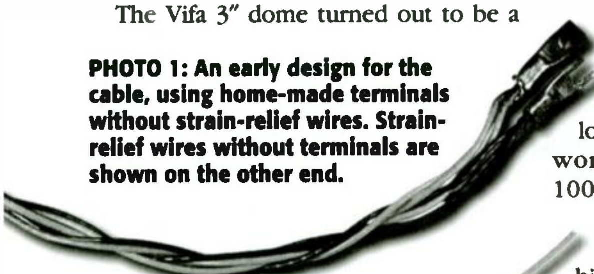


PHOTO 1: An early design for the cable, using home-made terminals without strain-relief wires. Strain-relief wires without terminals are shown on the other end.

PHOTO 2: Final design, with Radio Shack gold series 278-311A connectors soldered to Radio Shack automotive #8 Megacable for strain relief. Heater hose covers the connection to the six-conductor braided THHN #12 wires. To avoid tangles, two of the three wire pairs have been coiled prior to braiding.

INDUCTANCE CONSIDERATIONS

On the other hand, cable inductance was less of a problem as long as it was the same in both channels. Past experience with series networks suggests that it may be important to prevent high-order distortion products stemming from the woofer and its network from entering the mid and tweeter circuits. Keeping cable resistance and inductance low allows the amplifier to damp these distortion products.

It should be noted that the use of ferrite-core woofer-network inductors with a no-feedback amplifier will produce plenty of driver intermodulation at high power. With highly damped amplifiers, iron-core inductors may actually lead to lower distortion because of lower resistance and better woofer damping. I had no trouble finding these interactions with an old tube-model distortion analyzer saved from the scrap heap. Capacitance and dielectric effects proved, as expected, not to be problems. This led me to a cable design (*Photo 1*) that has very low resistance (0.064Ω for 60′) and fairly low inductance.

IT'S A LONG WAY

My cable run is 60′, allowing the CD player and amplifier to be in a dust-free area. The crossover is often at the listening position, so I can hear changes immediately. This dictates the use of #8 wire that is

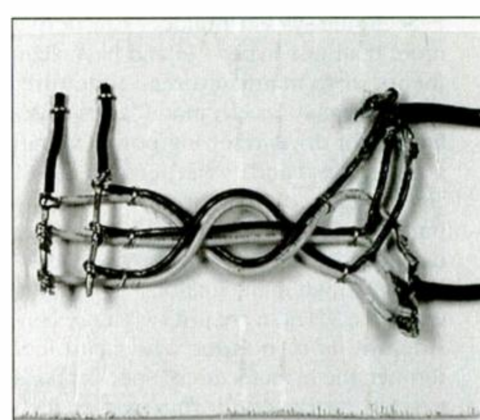


PHOTO 3: Exploded model of wiring and braiding technique.

reasonably flexible so I can easily move the crossover while operating.

Single-conductor #12 stranded wire of type THHN is available at low cost for use in conduits. Braiding six single conductors yields a two-conductor cable that is equivalent to #7 wire for as little as 27¢ per foot (*Photo 2* and exploded *Photo 3*). Once braided and terminated, there is no way the cable can unwind, and a moderate degree of flexibility results from the braiding. For a small additional cost, I used short lengths of Radio Shack #8 fine-strand single-conductor Megacable to provide color coding (red and black) and greater flexibility at the terminations.

TESTING TERMINATIONS

In an all-out war on resistance, I believe a good look at terminations is essential. Ironically, huge connectors don't offer any advantage over carefully soldered small, gold-plated Radio Shack lugs selling for under a dollar each. I tested all connectors for harmonic distortion and rectification as well as resistance, and I even checked frequency response. At worst, the behavior was equal to one inch of #12 stranded wire.

I partially ground down one Radio Shack gold lug to the copper base and put it in salt water for six weeks. The copper corroded where exposed, but no blistering took place, which indicated a good plating job. *Photo 4* shows the test cable I used to check terminations. I did not investigate gold binding posts, but I suspect the least expensive ones are a good investment if you're in a damp climate.

A small soldered terminal lug will outperform the largest crimp-on connector. If the small lug will not accept all the strands of a large wire, I simply solder the remaining strands to the outside of the lug (*Photo 5*). Note that the rubber boots supplied with the smallest lugs will not fit over Megacable. All the lugs I used were designed for crimping.

Soldering these is easy, provided you always use a vise to hold the wire end you're terminating straight up. This prevents solder from flowing onto the part of the lug that's in contact with the bind-

ABOUT THE AUTHOR

Jesse W. Knight has designed speakers for home and church use. He has installed sound reinforcement and recording systems for court houses. His work on a six mile submarine power cable installation involved separating submarine cable facts from fiction, resulting in an on time project that was within budget.

ing post. Solder flows onto gold very quickly. The key is to maximize the surface area of the connector-wire interface for a strong low-resistance connection. All rosin-core solders are equally effective whether they are lead, antimony, or silver. I believe the only important issue in choosing solder is that lead is toxic.

PHOTO 5: Details of wire connections: boots have been left out of position to show soldered connections to the gold terminals. All three of the blue THHN wires have been soldered to the black Megacable, but only one of the yellow THHN wires has been connected to the red Megacable to show how the Megacable strands are divided for easier soldering.

MECHANICAL CONSTRUCTION CONSIDERATIONS

Making the two Megacables different lengths produces a less bulky connection, since it staggers the solder connections. I divided the other end of the Megacable into three bundles of strands so I could solder each #12 wire separately. This requires a 40W large-tip iron. If you have a 100W iron, you can solder all three #12 wires at once. In *Photo 6* the connections are taped, and in *Photo 2* a piece of heater hose has been pushed over the connections for a neater appearance. Finally, I forced silicon rubber into the hose to keep it in place.

For the lowest inductance, you must braid three wire pairs as shown in *Photos 2* and *3*. If you are making a lot of cable, it is best to buy two 500′ spools of THHN #12 wire of different colors (blue and yellow in my case) to prevent confusion. Otherwise, you should mark the neutral wires with black tape. It is not necessary to braid the wires any tighter than shown in the photos.

PHOTO 4: A test setup used to test losses in connectors. The small twisted wires were connected to a high-gain oscilloscope to sample the voltage drop across six Radio Shack 278-316A and two 278-334 terminals.

WIRE TESTS

THHN wire is made for 60Hz use, so it is reasonable to wonder about dielectric effects at high frequencies. I do not have the equipment to detect every conceivable effect, but based on the testing I could do, I found no evidence for nonlinear effects. In the worst-case scenario, a speaker cable is driven by a nonfeedback amplifier with a 16Ω output impedance and loaded with a 16Ω speaker. In this case, any cable capacitance will be shunted by 8Ω. When you use an amplifier with feedback, the total impedance the capacitance sees drops to a fraction of an ohm.

My test exaggerates any potential dielectric effects by driving one end of the 60′ cable at 80V pk-pk through a 150Ω

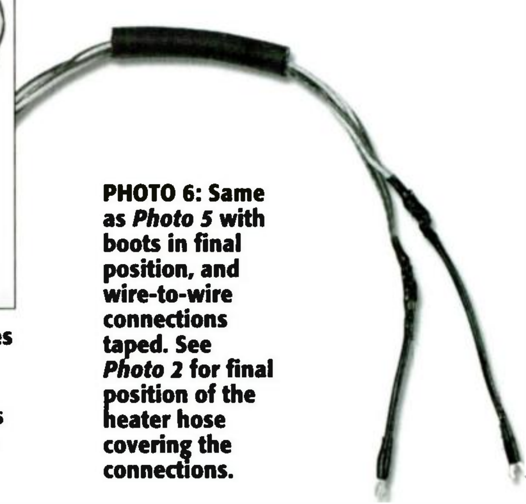


PHOTO 6: Same as *Photo 5* with boots in final position, and wire-to-wire connections taped. See *Photo 2* for final position of the heater hose covering the connections.

carbon resistor at 20kHz. The THD measured 0.09%, which is the floor for the test setup. Looking at the distortion analyzer output, I was unable to find anything except the signal-generator third harmonic and some AM radio interference, despite the fact that the cable was contained in a grounded steel garbage can with the lid on. Without the can, interference was so great that no meaningful measurements could be made.

This led me to ponder why people who worry about dielectric effects in speaker cables do not shield them in multiple coaxial copper water pipes grounded every foot. Regardless of drive impedance, broadband interference swamps dielectric effects. I had already learned this in working with microphone cables, but I just wished to double-check.

INDUCTANCE

My cable has more inductance than a ribbon cable, but the resistance is much lower. To achieve the same low resistance with ribbon cable would be expensive, and capacitance might actually be a problem with the large number of parallel wires. Ribbon cables are ideal for moderate-length runs under carpets. My 60′ cable has a loss at 60kHz of 0.4dB, which is less than the loss of the Hafler amplifiers used to drive the cable. Amplifier damping at audible frequencies is not seriously degraded by the cable.

AMPLIFIER LIMITATIONS

In designing the DH 200 and DH 120 amplifiers, Hafler put a premium on stability by decoupling the speaker load with a filter at supersonic frequencies. Needless to say, starting the high-end rolloff just above the audio spectrum is controversial, but I have found nothing to fault in these amplifiers. Series resistors in the preamp decouple its inputs and outputs as well. I have definitely heard the transient intermodulation and ringing that results from instability in other amplifiers that were "faster," and it is nasty. The effect of my 60′ cable on a 20kHz square wave is quite small compared to the effect of the amplifiers used. Given this, I don't see a need for a greater number of conductors in my cable for use with these amplifiers.

While this cable takes time to construct, it appears to completely fill my needs. Resistive losses in the 60′ length are so low that you can completely ignore them. Listening tests with various networks indicate that it is transparent for 4–8Ω loads.

Ask SB

A PAIR OF GEMS

As the owner of a pair of Geminis, I'd like to thank Mr. D'Appolito for developing this terrific configuration ("Test Drive the Dynaudio Gemini," *SB* 7/97, p. 38). Although it's now clear that the Geminis do not meet all of the design objectives in Mr. D'Appolito's 1983 AES paper, they do provide very good imaging and throw a wonderfully deep and wide soundstage. I, like Mr. Florian, was surprised at the apparent shortcomings revealed by the measurements. Several questions come to mind:

1. Do the ripples between 3–15kHz significantly affect the subjective sound quality (and how)? Can the addition of rounded corners to the cabinets minimize this effect? If so, what radius do you suggest?

2. What causes the 2.5kHz peak in the woofer? Can it be corrected?

3. What subjective effect should the "bizarre vertical polar response" cause? (I have noticed no problem in the stand-up-sit-down test on my speakers. Is there any chance the tweeters are wired backwards on the tested units? I found the crossover wiring a little confusing at first.)

4. Would a third-order crossover provide improved measured and/or subjective results? (Perhaps a modified or redesigned crossover could be the basis of a follow-up article.)

In short, I love the sound of my Geminis, but if there's room for improvement, I'm "all ears."

I would also like to suggest that future articles include more comments on the perceived sound, as this is the main reason for buying or building high-quality speakers.

Hugh James
Hamilton, Ontario, Canada

Joe D'Appolito responds:

1. The on-axis ripples are due in large part to diffraction off the vertical edges of the enclosure which arrive at the listening location about 0.3ms after the primary wave off the driver diaphragm. This can lead to a subtle smearing of sound. Almost all loudspeakers exhibit this level of diffraction. In the Gemini, the ripples decrease rapidly as you move off-axis in the horizontal plane.

You can answer the question yourself by comparing on-axis sound with slightly off-axis sound. You must be careful, however. The farther off-axis you are, the less diffraction effect, but then you must contend with the tweeter high-frequency fall-off.

There are only two ways I know to produce a truly diffraction-free system: get a true point source or place all drivers flush on an infinite baffle (or at least on a very large wall). For edge rounding to be effective in reducing diffraction, the radius of the round must be comparable to a wavelength. At 3kHz this works out to a radius of about 4.5".

2. I did not measure the individual driver responses in the Gemini. The peak is probably due to the woofer, but without this data I cannot say with certainty where the peak is coming from or whether and how it can be corrected.

3. The "bizarre vertical polar response" is best heard by placing the Gemini on its side and moving in a horizontal arc. You will hear it.

4. In practice, odd-order crossovers do not work well with the MTM geometry, notwithstanding my paper. The problem is caused by the time offset between the tweeter and woofer pair when mounted flush on the same baffle. This requires a greater than 90° phase shift between drivers to get flat response. Dynaudio

tried to get around this with an all-pass network. This leads to a very linear phase response on-axis (which is good), but also messes up the vertical polar response (which is bad).

All designs involve compromises and trade-offs. Dynaudio made its choice. If I designed a system with these drivers, I would use higher-order (second or fourth) in-phase crossovers and give up linear phase for better polar response.

If you love the sound of your Geminis, don't let a few measurements scare you off.

SWAN'S SONG

Back in 1988, an article on the Swan IV Loudspeaker (Issue 4/88, p. 9) by Joe D'Appolito was published in *Speaker Builder*. Having heard a Swan IV loudspeaker system that a friend constructed from that article, I am now interested in building a set.

One of my concerns is the availability, or, more properly, nonavailability of the electronic crossover. Having seen an article on the XVR-1 Electronic Crossover in *SB* 2/96 (p. 18), I wondered whether it would serve as the electronic crossover for the Swan IV, especially since the XVR-1 can provide bass boost for such sixth-order bass reflex woofers.

Both the Swan IV and XVR-1 appear to be exceptionally well designed audio components. Will they work well together?

Also, since two 100-ltr enclosures are too large (wide and deep) for my listening room's decor, is there a way that the Swan IV bass enclosure could be made smaller while maintaining low frequency cutoff but sacrificing some SPL output?

Finally, have any improvements been made to the Swan IV satellite speakers?

M. L. Piccione
Hazelton, PA

Joe D'Appolito responds:

With regard to the Swan IV electronic crossover, the XVR-1 is an excellent substitute for the original Pedal Coupler. It will provide the necessary low-frequency boost plus the 200Hz crossover. The high-pass crossover is first-order; the low-pass is second-order Linkwitz-Riley. The woofers must be connected with reverse polarity.

As I explained in my original article, the optimum enclosure volume for the two 12″ woofers is 140 ltr. We already reduced this to the minimum of 100 ltr. If you require a smaller box, you can use just one 12″ unit or perhaps two 8″ or 10″ units. In any event, the woofer needs to be redesigned. A pair of 8″ Focal 8V416Js will fit in about 60 ltr.

There have been no formal updates to the Swan IV. The design is now over ten years old. Were I doing a Swan IV today, I would certainly explore newer drivers. That said, the Swan IV is a classic design that still competes well against today's designs.

HOW TO BREAK IN SPEAKERS

Would someone please advise me on the art of breaking in a new loudspeaker—the best frequency, amplitude, and length of time it takes to get the beast "settled in"?

A.J. Steen/Renkon
Hemet, CA

Dennis Colin responds:

Since I haven't purposefully compared sound before and after break-in, I can't comment on recommended amount. But since the purpose is to "exercise" all of the speaker's components, you should use a full-spectrum signal such as pink noise or wide-range music.

Pink noise is good for consistency, but should be slightly tilted down toward the upper end, since, compared to woofers, tweeters can usually handle less than 10% of the power. In *SB* 3/99, pp. 38-39 ("Swans M1 Kit Review"), I used tone controls to achieve about 8dB less spectral density at 20kHz than at 20Hz, about a 6.3 to 1 power ratio. Note: do *not* use white noise, such as from an FM tuner between stations. White noise has flat power per Hz bandwidth, which means 500 times more power in the top octave (10–20kHz) than in the bottom (20–40Hz). The de-emphasis in FM tuners lowers this somewhat, but chances are if you crank it up to see the woofer cone move, you'll see tweeter smoke first!

Any music with a wide frequency range is good for break-in; the question is how loud. I recommend connecting an AC voltmeter across the speaker(s) and allowing about 10V peak readings; this would be 25W across 4Ω. Of course, the average power is much lower, so there should be no risk to even small drivers. Yet 25W peaks will move the woofer a surprising amount, not to mention your ears!

Regarding the latter, in my Swans M1 review I didn't wish to hear loud pink noise for two hours, so I butted the two speakers together front to front, and drove them in opposite phase. This canceled much of the sound and also allowed the greatest woofer cone motion—about ¼″ peak-peak with only 1W average noise per speaker.

Here's my proposed simple test to determine both audibility and sufficient duration of break-in:

1. Alternately, listen to each of a pair of speakers with the same signal and with the speakers next to one another to make sure they're well matched.

2. If so, then break in one unit for a short time, say a half-hour.

3. Repeat #1. If there's now a noticeable difference or more than before break-in, repeat #2 and #1 until there is no further change in this difference.

4. Then break in the other unit for the same total time.

Of course, this takes more than twice as long, but you will have performed a valuable research experiment!

PORTS: HOW LARGE?

I have worked out a design and have determined a range of vent-tube lengths and corresponding diameters. What I'd like to know now is whether there is some worked-out theory to determine an upper bound on the area of the vent? I've heard a rule of thumb (for a circular cross section) that the diameter should be no larger than three-fourths of the speaker's diameter. But I would like to have a more analytical basis—probably involving the Helmholtz frequency (f_B), certain box dimensions, and so on. I believe the question comes down to: at what point does the vented box stop acting like a Helmholtz resonator?

Eliot
Eliot@mathtechinc.com

Dick Pierce responds:

This is actually a very interesting problem. Almost all work that's been done on the proper sizing of vents has concentrated on determining the minimum vent size. This is because of the increased particle velocity in the vent as the diameter becomes smaller and the attendant nonlinear effects that might result.

However, very little consideration has been given to what might be a reasonable upper limit to vent diameter. The problems at this end are sometimes a little more subtle, and sometimes obvious. Let's look at some of those issues.

For a Helmholtz resonator to work requires two reactive acoustical elements: an acoustical compliance, supplied by the volume of air in the box; and an acoustical mass, supplied by the port. As you may know, the acoustical mass in the port is directly proportional to the length of the port and inversely proportional to the square of the port's diameter. This is why when you increase the diameter of the port, you must increase its length to maintain the same acoustical mass.

But to behave as an acoustical mass, the air in the port must essentially be moving as one. Implicit in this is the requirement, then, that all port dimensions must be substantially smaller than a wavelength of sound. As one of the dimensions starts to approach a wavelength, the port no longer acts as an acoustical mass, but begins to exhibit the behavior of an organ pipe or even a transmission line. Not all the air is moving in the same direction, and the port no longer behaves as a single mass.

So, clearly, we have a set of limits imposed by the requirements that the port act as a single lumped acoustical element. The diameter must be small enough so that it is a tiny fraction of a wavelength in the region of the Helmholtz resonance. At the same time, port length resulting from the diameter must also be a tiny fraction of the wavelength at these same frequencies. Using an arbitrary notion of "tiny" as less than 1/10 of a wavelength, this would suggest that the largest dimension of a port operating at 50Hz (wavelength is 20′) would be a 2′.

I would, personally, adopt an even more conservative figure, such as 1/15 or 1/20 the wavelength, suggesting that a port tuned to 50Hz should not exceed about 16″ in length. This is because the port is not just acting at 50Hz, but over a substantial range of frequencies above and below this. Imagine, also, that a 2′ long port will exhibit its first standing-wave resonance at the low frequency of only 250Hz, where the output of the port increases substantially.

A second consideration is that to act as an acoustic mass, the port must form an obvious "tube" connected to the enclosure. This might seem overly obvious, until you consider that if the port cross section had the same area as one wall of the enclosure, it would simply look like an increase in the enclosure size, not a port adding an acoustical mass to the system. So another restriction on the maximum diameter of the port is based not on the driver's diameter, but the physical dimensions of the enclosure: to act like a port, it must look like a port.

A third consideration is one of real estate. As we mentioned, the larger the port diameter, the longer the length must be for a given acoustic mass. After a point, the length will be long enough that you now have a problem finding where to put the port.

At the other end of the limit–consideration of the minimum diameter–I suggest that things are not as neat and tidy as some of the literature might suggest. Usually, as mentioned, the criteria used is to limit the particle velocity to some small fraction of the speed of sound to limit potential nonlinear motion. However, I don't recall either a rigid theoretical study nor careful measurements confirming the existence and magnitude of these suggested effects.

Further, the maximum particle velocity will occur for a given SPL only at the Helmholtz resonance Fb, and is reduced on either side of that. It is at Fb, remember, where the port is producing most of the system's acoustical output, and the output from the driver is at a minimum. Outside of the range of Fb, more and more of the system's output is from the woofer, and less from the port. Rest assured that the nonlinear behavior of even the best woofers is far worse than all but the tiniest of ports at the kinds of sound pressure levels where the port velocity might be an issue.

So, to summarize the answers to the questions you posed: make sure the diameter of the port is small enough so that the largest resulting dimension is substantially smaller than the wavelengths over which the port is expected to operate as a port; make sure the port is not so big so that it merely appears as an extension of the cabinet volume; and make sure the port is small enough to fit in the enclosure. The first two criteria are the result of considerations of the physical behavior of ports, while the third is a practical construction issue.

ARIA UPDATE

With regard to the Aria 5 speakers ("Test Drive," *SB* 6/98, p. 44; see also "Kit Report," *SB* 4/92, p. 40), first, let me express my greatest appreciation to *SB* and to Mr. D'Appolito for his design of the Aria 5 and other related speakers. I started to build a set of Aria 5 speakers in 1991, but due to my busy schedule did not finish them until about 1½ years later. I have been greatly enjoying the speakers ever since.

I built the speakers to the published specifications, but made a few minor changes to the enclosure, making it out of MDF, and then applying a hardwood veneer for a beautiful finish. I also built matching stands, which produced a very elegant-looking speaker system. But the most noticeable addition was a small internal enclosure around the tweeters.

During the initial listening tests, I noticed that

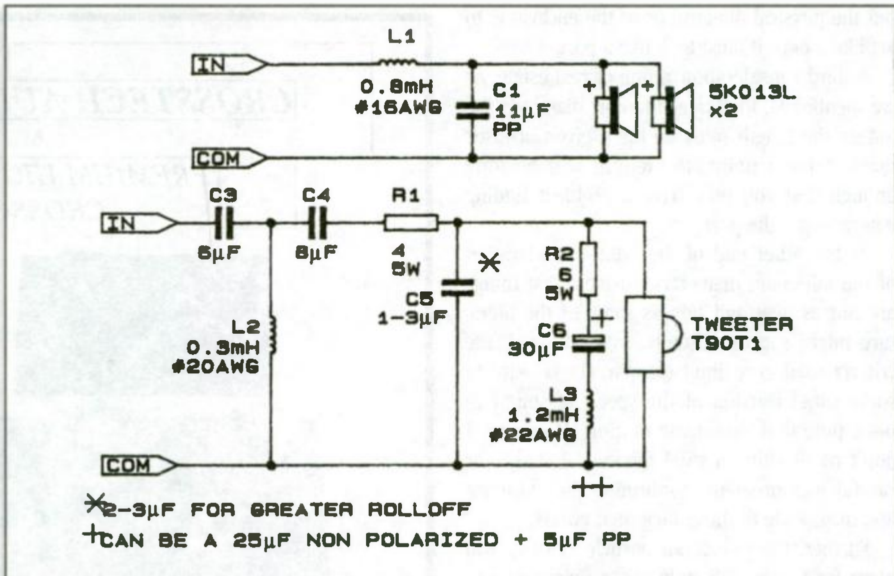


FIGURE 1: Aria 5Ti crossover. B-asksb-1

the plastic face plate of the tweeter vibrated severely at certain resonant frequencies. The vibrations could be easily felt by placing my fingers on the front of the face plate, and were quite severe when listening to recordings with strong tenor vocals. I used ⅜" Plexiglas® to create a small enclosure behind the tweeter, which took up negligible volume within the cabinet. The tweeter enclosure eliminated the noticeable vibrations in the tweeter face plate, and greatly improved the high frequency sound quality and imaging.

Shortly after I purchased the original drivers (T90K and 5K013L) and crossover components, I heard about the titanium T90Ti version, but decided to complete what I had started. Like other readers, I have noticed the slight harshness in the TK90 tweeter, and have been interested in upgrading to the T90Ti. A few years ago I purchased some T90Ti drivers, but am just now getting around to upgrading my Aria 5 speakers.

After searching through the last several years of *SB* to find the Aria 5Ti crossover, I noticed that there has been little or no mention of the Aria speakers in recent years. Are the Aria speakers considered to be totally "out-dated?"

In preserving Mr. D'Appolito's years of research and the quality of the Aria, I wish to maintain the exactness of the original Aria designs. Thus, I am facing some questions, and hope that you can provide the answers.

1. I have included the crossover schematic (*Fig. 1*) from *SB* 1/93 ("*SB* Mailbox," p. 62). Is this the correct crossover for the Aria 5Ti? If not, what is the latest version?

2. Why has the bass equalization to correct for diffraction loss been removed in the Aria 5Ti? And what are the effects?

3. Without the bass equalization, does the Aria 5Ti need a subwoofer? I have been very satisfied with the bass response of the original Aria 5, and did not plan to add a subwoofer. Does the Aria 5Ti have as adequate, or should I say, surprisingly sufficient, bass response as the original Aria 5?

4. How does the absence of the base equalization affect the integration with a subwoofer such as the Aria 10?

5. What type (or specific model) of subwoofer is recommended for the Aria 5Ti? What crossover frequency and slope are recommended?

6. In *SB* 6/92 ("*SB* Mailbox," p. 58), Mr. D'Appolito included a frequency-response plot of the Aria 5Ti down to 200Hz. What does the frequency response look like below 200Hz? Is a frequency plot of the full frequency range available?

7. In *SB* 5/92 ("*SB* Mailbox," p. 51), Mr. D'Appolito stated that the crossover was simplified to reduce phase shift. What are the trade-offs, or disadvantages, in trying to add the diffraction loss equalization to the Aria 5Ti?

8. What is the crossover for the Aria 5/Accuton with the Accuton C2-11? Is this design more current, and how superior is it to the Aria 5Ti? Is there a more current version of the Aria 5?

9. Has Mr. D'Appolito experimented with expanding the mid-tweeter-mid configuration to in-

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clude the subwoofers for a woofer-mid-tweeter-mid-woofer configuration? Is this idea worth exploring? What are the advantages and disadvantages?

10. I have not seen any ads for the Focal 5K013L or T90Ti drivers in some time. These drivers are excellent. Have they been discontinued or replaced? I realize that there could not be a direct replacement for these drivers in the Aria 5 designs (luckily, I have already purchased the drivers), but has Focal come out with a higher-quality replacement for these drivers that could be used in similar designs?

Again, many thanks to Mr. D'Appolito and *SB* for the excellent work and for the assistance that you provide to your readers. I am excited about getting back into experimenting with and building speakers.

Paul Smith
Plano, TX

Joe D'Appolito responds:

The responses below are numbered to correspond with your numbered questions.

1. This is the most recent crossover.

2. The original Aria 5 crossover accomplished the crossover function and diffraction loss with two circuits. Later analysis showed that a simple change in the crossover Q could accomplish both functions with a substantial reduction in the number of parts.

3. The Aria 5 and the Aria5Ti have been designed as near full-range stand-alone systems. Bass response is the same as in the original and extends down solidly to the mid-40s.

4. See 2 above.

5. I do not recommend a subwoofer with the Aria 5. I would use a closed box with the same drivers if you plan on a subwoofer.

6. See 3 above.

7. This question is based on a misunderstanding of the latest crossover. It is now moot.

8. I do not have a copy of the Aria5/Accuton crossover in my files. I believe the C2-11 has been replaced with the C2-12. If I were designing an Aria5/Accuton today, I would use the C2-23.

9. I see little advantage to a three-way W/M/T/M/W arrangement unless the woofer-to-midrange crossover is quite high (800Hz or more). The upper woofer is less than optimally loaded, so bass power is sacrificed. Lobing error is much less of a problem at more typical crossover frequencies in the 300–400Hz range.

10. The 5K013L and T90Ti are alive and well. There is now a TC90TDX also.

SB Mailbox

CORRECTED CROSSOVER

The schematic for the diamond crossover (*SB* 5/99, "The Diamond Nearfield Monitor," p. 36, Fig. 3) has a mistake in it. The R2 value is shown as 16Ω; it should be 1.5, or 1R5Ω.

Philip Abbate
Duluth, GA

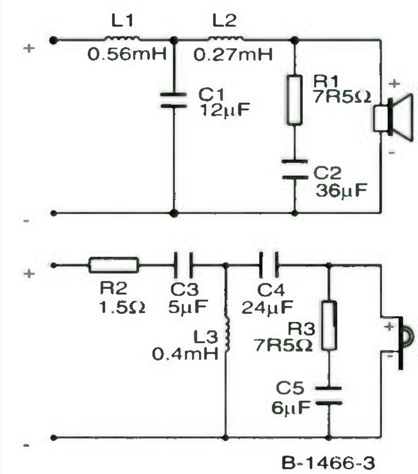

FIGURE 1: Corrected crossover.

THREAD OF ANALYSIS

In *SB* Mailbox (*SB* 4/99, p. 53), Mr. Jenkins has expanded his argument with some new published data that the speed of sound in fiber does not change. He also addresses the question in a form that I'm the opponent in the debate: "If Mr. Jakulis has a problem with the experiment not producing the results he wishes, he can question the competence of the person making the analysis."

I wish that we could have communicated directly and perhaps avoided the apparent misunderstandings. The issue is not my beliefs but data that has been investigated by widely accepted academic researchers, such as R.H. Nichols, "Flow Resistance Characteristics of Fibrous Acoustical Materials," *J. Acoust. Soc. Am.*, vol. 19, #5; R.A. Taub, "Fiber-Tangle Interaction with an Airflow," *J. Fluid Mechanics*, vol. 27 pt. 3; Z. Esmail-Beguin and T.K. Naylor, "Measurements of the Propagation of Sound in Fiberglass," *J. Acoust. Soc. Am.*, vol. 25 #1; and L.J.S. Bradbury, "The Use of Fibrous Materials in Loudspeaker Enclosures," *JAES*, April 1976, vol. 24 #3.

Also consider the derivative results as

pertaining to the TL: R.M. Bullock, "A Transmission Line Model Woofer Model," *JAES* 2384 (D-6), and J. Backman, "A Computational Model of Transmission Line Loudspeakers," *JAES* 3326(2TD1.12), where the concept of the change of speed of sound in the fiber mass is central to the argument. Thus, Mr. Jenkins' hypothesis that "there is no delay in the pulse transmission time due to wool filling" is at odds with the above data and not with my beliefs.

I have pointed out some inconsistencies in the data that Mr. Jenkins used to buttress his argument; however, I hope that I have not questioned his competence or integrity. Consider Mr. Jenkins' statement that "The most interesting commentary comes from readers who 1) don't for the moment believe that your data is any good." Let me offer the following observation of the apparent inconsistency in the Fig. 5 data that Mr. Jenkins has offered as proof.

For a 50″ (1.24m) line, the fundamental resonance would be about 68Hz, thus it would not support a 40Hz signal. The Fig. 5 data does not correspond to a TL response, and with a wool fiber density of 0.49 lb./ft^3, the attenuation would be greater than 15dB. This is not the case with the shown data, unless the signals are of a different scale which is not made clear. Thus the conditions under which the data was obtained are not explained and the data is apparently inconsistent with the fiber mass attenuation as well as with the change of speed of sound in the fiber mass.

In this debate my view is that Mr. Jenkins has a double standard to meet: not only must his data be convincing, but he must also show where the above-mentioned authorities have made the mistake in the conclusions reached. A comprehensive analysis of the TL would be the icing on the cake since disregarding the speed of sound change resulting in the frequency shifts of the line harmonics would make the analysis of the TL so much simpler.

E. Jakulis
Avon, MA

Don Jenkins responds:

Mr. Jakulis has correctly pointed out that a number of "widely accepted academic researchers" have decided that a fiber-filled transmission line reduces the internal sonic velocity from that of free air. I can not disagree with that observation.

On the other hand, I can point out that over the past year of making an extended series of experimental evaluations trying to prove this "widely accepted fact," I was unable to do so. My final test configuration used in this series followed a minimalist approach by using only time and distance as the analytical parameters. After starting with resonant mode and microphone probe techniques, I came to the conclusion that resonant mode analysis, including microphone probe methods, which rely on resonant, or standing wave, modes, may be misleading in the interpretation of the measurements. Obviously there are others, almost exclusively I might add, who disagree with this observation.

I can only offer my test results for review and leave the acceptance or rejection of the results and conclusions to the reviewer. These latest results are contained in a new report that may be published by *Speaker Builder*. A summary of these results can be stated as: Three different experimental methods were used to measure the transit time for a compressive wave front to move through a heterogeneous medium of air and fiber, all using a direct time difference and distance measurement. Wool and Acousta-Stuf, both at the magical loading density of 0.5 lb/ft^3, were evaluated. In none of these cases was any significant reduction in the transmission velocity measured (all within 3% of free air velocity).

I have supplied Mr. Jakulis with the full report of these tests and asked for his commentary. I have also proposed that he configure a test, using the direct time of transit technique, that will show the anticipated reduction in velocity. If the "widely accepted fact" of velocity reduction is indeed a fact, then it should be easy to demonstrate this with a simple direct time of transit and distance measurement.

In response to Don Jenkins' letter (*SB* 4/99, p. 60), I'm attempting to explain the differences in our experimental results. I have measured what I consider to be the steady-state sound velocity in a lossy acoustic (stuffed) transmission line (*Table 1*). I use this data in a transmission line simulation program that plots SPL, which presents a steady-state view of the speakers' performance. In Jenkins' article (*SB* 7/98), he measured the resonant frequency of a stuffed pipe to determine the velocity of sound. The resulting velocity he obtained was steady state and was in substantial agreement with my results.

I have included a diagram of my latest test arrangement for measuring phase differences using Lissajous figures to identify 90 and 180° phase shifts (*Fig. 1*). I measured the wavelength of sound in the "stuffed" line and have calculated the sound velocity from that and the sound's frequency.

TABLE 1
STEADY-STATE VELOCITY MEASUREMENT ON STUFFED LINE

Freq.	Mike Pos1 Inches	Mike Pos2 inches	Phase diff. degrees	Pos2 – Pos1	Wavelength Calculation	Vel. Cal.
40.0Hz	13⅜,	46	90	32⅝,	130.5	435.1
56.6Hz	7	45½	120	38½	115.5	544.8
80.0Hz	20⅝,	43	90	22⅜,	89.5	596.8
113.1	12¾	30⅛,	90	17⅜,	69.6	655.5
160.0Hz	20½	33½	90	13	52.0	693.3
226.3Hz	24⅛,	34¼	90	10⅛,	40.5	763.8
320.0Hz	24⅞,	39⅞,	180	15	30.0	500.0
452.5Hz	19½	31¾	180	12¼	24.5	923.8
640.0Hz	23½	32¼	180	8¾	17.5	933.3
905.1Hz	17	23⅜,	180	6⅜,	12.76	962.4
1280.0Hz	7⅝,	12¼	180	4⅝,	9¼	986.7
1810.2Hz	8⅛,	12	180	3⅞,	7¾	1168.7

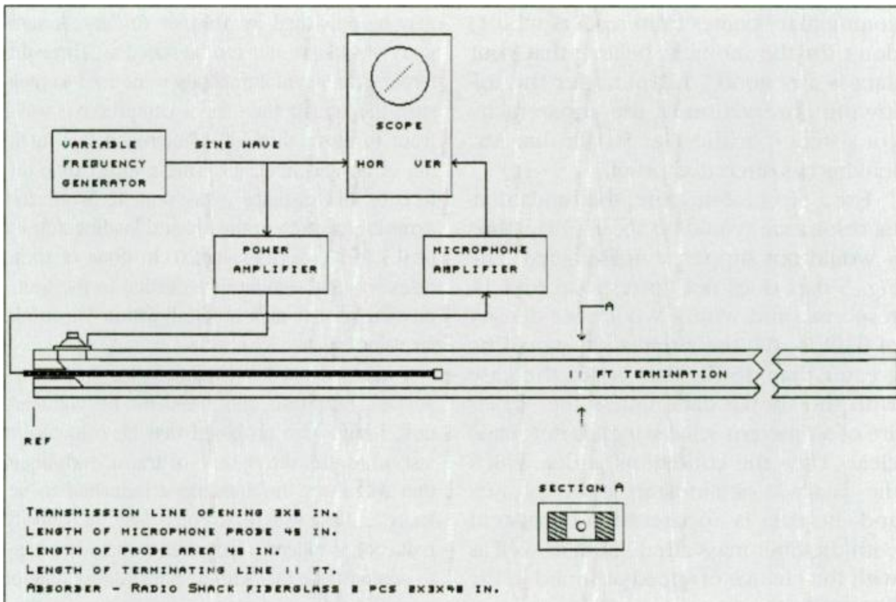

FIGURE 1: Author Mattern's test setup.

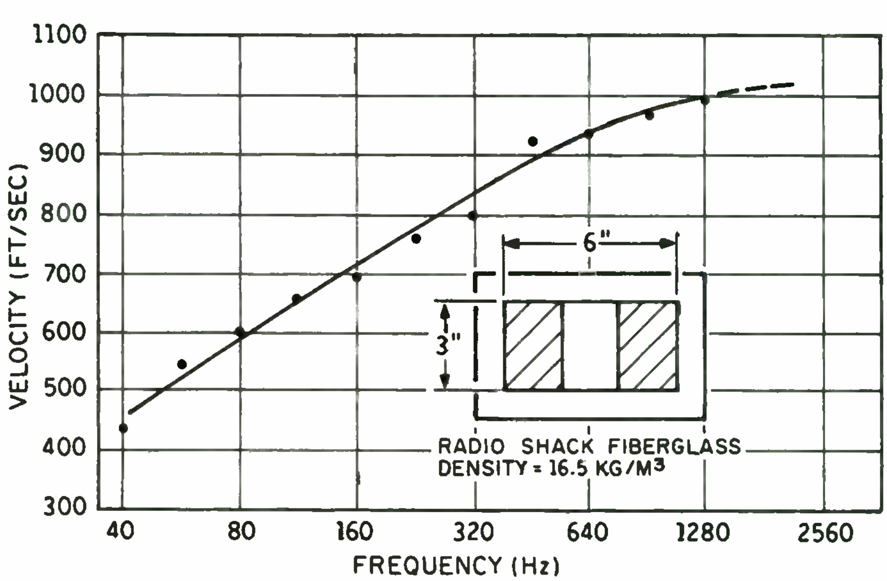

FIGURE 2: Plot of measurement results.

Test data and a plot of the results (*Fig. 2*) are also included. As you can see from the plot, the stuffing causes a very significant reduction in steady-state velocity at the bottom end of the audible spectrum. Also, the plot shows the steady-state velocity to approach the velocity of a bare line at high frequencies. At high frequencies the velocity is unimportant as the sound is highly attenuated.

In Don Jenkins' second letter (p. 54), he talks of testing with a single 50Hz sine pulse. A 50Hz sine pulse has a continuous spectrum extending from DC to over 100Hz. This does not provide a 50Hz steady-state forcing function.

In another test he uses an exponential burst of 10 cycles at 40Hz. This is a little like using a single sine pulse since the stuffing is reacting to the previous weak pulses while the mikes are comparing the last stronger pulses. This is a step—but not good enough—toward a steady-state comparison. Why not use ten or more pulses of equal amplitude?

John Mattern
Baltimore, MD

Don Jenkins responds:

Mr. Mattern has asked two questions regarding the letter that appeared in *SB* 4/99. The first question asks why use a single sine pulse of 50Hz rather than a continuous wave.

The object of the tests being reported was to determine the velocity of sound in the fiber-filled tube by direct measure of the time of transit of the compressive wave front over a specific distance. Sound is a change in pressure. The velocity of the initial compressive wave, seen as a pressure rise of variable amplitude and rate, is the parameter to be measured. For the tests described, I used a microphone, which is a sensitive pressure transducer. Since only the velocity was of interest, additional cycles were not needed.

The second question is about the use of 10 cycles at 40Hz, increasing in amplitude during the pulse train. A part of the Bradbury theory that describes how the velocity of sound is reduced by the fiber in the sound transmission path requires that the fiber mass be in compressive wave coherence with the air. This means that the individual fiber "tangles" accelerate and decelerate along with the air molecules. This is how Bradbury develops the increase in bulk density for the fiber-loaded line.

Since the inertia of the fiber is several orders of magnitude greater than the air, it may be assumed that a single pulse transmitted through the filled medium will not set the fiber in oscillatory motion. This is the reason the Bradbury theory shows that the velocity reduction is more effective at low frequencies than high frequencies. At the higher frequencies, the inertia mass is simply too high to allow synchronous oscillation with the air.

The thought behind the 10 cycles, increasing in amplitude with each cycle, was to allow the fiber to come to phase equilibrium with the air, a condition required for the Bradbury theory to be effective.

Both questions ask why not let the system come to equilibrium by using more cycles. Once the system starts to reflect transmitted energy back to the source, the initial wave-front positions will shift to new positions depending on the transmitted frequency, the length of the line, the closure characteristics of the line end, and the attenuation of the reflected energy due to the fiber. Since the object of the test was to only determine the initial time of transit, more cycles were not needed.

TL DESIGN

I was very interested and pleased to see John Mattern's Tapered Line design in *SB* 3/99 (p. 28), particularly since I was bemoaning the fact that I had seen no other designs since David Weems' in '87. Mr. Mattern's design is an intriguing variation on the partitioning, especially if you prefer a more compact enclosure. But my inclination is to have as few bends in the line (hence single partition) as possible, thus allowing the pressure waves a smooth flow within the enclosure. As with previous designs, the taller boxes eliminate the need for stands, but they still need spiked feet. My previous designs have all vented to the floor via a port of the area of the bass cone.

You have probably seen Castle Acoustics' commercial realization of this type of loading. Their more recent designs have the ports vent via a plinth with a small gap all the way round. I was intrigued by this and made several inquiries.

Tony Seaford of Marton Music, who has designed and manufactured many speakers, was very helpful. Briefly, research showed that measured and auditioning tests revealed that improvements were gained by having a resistive termination for the port. The gap between the cabinet and plinth on the Castle Howard is 3.5mm. Tony Seaford suggests restricting the final area to approximately 10% of the cone area.

I have tried this on previous designs and found it very beneficial (certainly with the particular drivers). The bass was at least as deep but tighter and better defined, and mid seemed clearer and sweeter.

So the above and forthcoming projects have ports on the bottom panels equal to the cone area, which vent via a plinth. In order not to restrict the gap too much (10% cone area), the plinths have their exit area on the sides, allowing gaps of only around 6mm, which I think is better than being too narrow!

R.D. Lewis
Llandeilo, Wales

John Mattern responds:

Mr. Lewis questions the impact of the numerous bends in my TL design. I have traded simplicity of construction and power handling for size in my design. Construction difficulty is the major disadvantage. Early tests with 3″ pipe elbows indicated that there was little, if any, effect of bends at low frequencies.

One of the objects of my design is to cross over to the cone's rear at roughly 60Hz. The taper increases the loading on the back of the cone making the port radiation stronger, but it also makes the crossover more difficult to achieve. The design requires an acoustical crossover with substantial attenuation at and above the frequency where the port is out of phase with the wave from the cone's front (approximately 200Hz).

The bends might actually be helpful to the extent that they discriminate against 200Hz compared to frequencies below 60Hz. This effect is small at best as the predicted and measured 200Hz dips are nearly equal. The simulation program neglects the bends.

I can assure Mr. Lewis that reflections at the bends are not coloring the sound. Pressure response with the mike very close to the cone is smooth and flat from the mid base to the woofer cut off. Most unevenness is coming from without the cabinet as a result of floor, wall, and ceiling reflections. Also, there is the usual loss caused by diffraction around the narrow enclosure. In the last version, the effect of wall reflections is minimized with absorbers beside the speakers, and the diffraction loss is compensated in the crossover network.

A large bass driver in a large box may not need to use the back wave to achieve deep bass. In this case, a TL that absorbs all the back wave may produce the best result. [But then, by definition, wouldn't the design be an infinite baffle? –Ed.]

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1993 Waslo's IMP, Pt 1-3 • Quasi-Monotonic Vented Alignments • Making You Room Hi-Fi, Pt 2-3 • A&S Soundoff Winner, Pt 3 • Flexible Dipole Woofer • Th Simpline • Stalking F_3 • A Bi-Structural Enclosure • A Sixth-Order T/S Subwoofer Design Speaker Enclosure Screws • Electric Bass Tri-Horn • Prism V Satellite/JBL Subwoofer, F 1-2 • Fitduct: Program for Designing Duct Software • Compact Coincidental Point Sourc Speaker • IMP: Measuring T/S Parameters • KIT REPORT: Rockford's Beginner Soft ware/Driver Paks • SOFTWARE REPORT: Low Frequency Designer 3.01 • Three Affordabl Measurement Microphones • Two Ways to Realize a Dream • Matching Driver Efficiencie • Two-Woofer Box System • Designing a Dual Voice Coil Subwoofer • SOFTWARE RE PORT: Blaubox 1.2 • Tale of Three Speaker Projects • A&S Sound-off 1992 • Monolit Horn • Orbiting Satellites • Real-World Three-Way Crossovers • The Simplex • Living wit a Speaker Builder • The IMP Goes MLS •

1994 Sanctuary Sonics • Modular Active Crossovers • A Full-Range Open-Baffle System • An Evolving Magnepan MG-1 • Low-Frequency AC-To-DC Converter • A Compact Bass Guitar Speaker • Measuring Speaker Impedance Without a Bridge • The Dynapleat • The Danielle II • The Birdhouse: A Sound-Reinforcement Subwoofer • The Linear-Array Sound System • A Revised Two-Way Minimonitor • Exploring the BUF 12 with Pspice • Signet's SL280B/U • Time Response of Crossover Filters • Converting Radic Shack's SLM To Millivolt Use • Acoustic Distortion and Balanced Speakers • Microphone Response Correction with IMP • More About the Birdhouse Bandpass • A 16Hz Subwoofe • D.H. Labs Silver Sonic Cables • The System III Loudspeaker • Exploring Loudspeake Impedance • IMPcycling • The Linear-Array Chronicles • Book Report: Loudspeake Recipes, Book One • The Woofer Test • A Large Ribbon You Can Build • Loudspeakers, A Short History, Pt 1-2 • Absolute SPL Sensitivity Measuring with IMP • The Damping Factor: One More Time • Cliffnotes for Loudspeaker University • Software Report: The Listening Room for Macintosh • Book Report: The Theory and Design of Loudspeake Enclosures • A 15" Transmission Line Woofer • Inductor Coil Cross Talk • Quick Home Theater on a Budget • Silk Purses: A Two-Way Salvage Design • Audio Phase Inductor •

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Book Review

THE INVENTOR OF STEREO: THE LIFE AND WORKS OF ALAN DOWER BLUMLEIN

Reviewed by Barry Fox

The Inventor of Stereo: The Life and Works of Alan Dower Blumlein, by Robert Charles Alexander. Focal Press. 421 pp., £29.99, $56.95 (available through Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, 603-924-9464, (Fax) 603-924-9467, custserv@audioXpress.com).

In the early 1930s, EMI's Laboratories in London, and Bell Labs in New Jersey, were working on hi-fi and stereo recording. Arthur Keller was the driving force at Bell, and Alan Blumlein pushed the boundaries at EMI. In those days, before electronic publication made the world a global village, neither team knew what the other was doing.

Neither spared a thought to the fact that what they were doing was commercially pointless, either. Worldwide unemployment meant that few people could afford one lo-fi loudspeaker, let alone two. But each lab was a hothouse for ideas, with far-sighted management able to see that today's blue sky research earns tomorrow's revenue.

THE EARLY YEARS

Alan Blumlein had started working in 1924 with International Western Electric (ironically a division of Bell Labs) and he stayed with the company for five years, developing electrical measurement and

to page 58

Speaker Builder Back Issues Continued

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Two pair Dynaudio T-330D Esotar tweeters, new, unused, tested for purity, $175/ea. Includes shipping. Daryl, (530) 265-2575, E-mail datara@yahoo.com.

Denon amp 200W/ch, $400; Harman Kardon PT2500 tuner preamp, $200. Both for $550, both in mint condition. Nick (559) 583-6511.

New: Vifa M22WR09 8″ woofers, $49/ea; Vifa P13WH 5″ drivers, $19/ea; Motorola 4×10 horns with KSN1086 drivers, $16/ea; Sledgehammer steel core inductors, $40/10; air core inductors, $19/12; Vifa tweeter grilles, $2/ea; MG45 metal grilles for 5¼″ driver, $3/ea. Used: SEAS P1REX 8″ polypropylene woofers, $15/ea. Rick, (610) 693-6167, audioarts@enter.net.

Two Scan-Speak 21W8551 woofers tested only (will provide T/S parameters), $65 each; two Audax Ti100DT $10/pr, used; two Allison 2.5″ dome mids, $40, O.B.O. (old handmade design); seven Dayton 5¼″ shielded midbass woofers $60 all. Contact snicson@aol.com.

Pioneer PDR-99 CD recorder/player/transport, list $2000, as new, in box, $595; Draper Envoy home theater motorized movie screen, 120″ diagonal, 1.3 gain, $500; JBL 2418H 1″ horn drivers, $200/pair; JBL 3110A 800Hz crossovers, $175/pair; Counterpoint SA20 amp, $800; SA20 with SA220 update, $1050. David, (914) 688-5024.

telephony equipment, while it mutated into the International Standard Electric Corporation and then Standard Telephones and Cables.

During this time Blumlein filed several patents, establishing a routine which thankfully means that although he wrote very few articles or technical papers, a total of 128 patent specifications gives us permanent access to Blumlein's original thoughts. The glory of patent law is once a patent has been granted, no one can change the wording. So mental processes are frozen in time.

By 1929, at the age of 25, Blumlein had become bored with telephony and joined the research team of the Columbia Graphophone Company led by Sir Isaac Shoenberg. His joining brief was to find a way round the Maxfield and Harrisom patents on electrical recording owned by Bell Labs. Another British company, HMV (The Gramophone Company), was also looking for ways round Bell's monopoly.

Blumlein cracked the problem, and went on to design a completely new electrical recording system. In 1931 The Gramophone Company and Columbia Graphophone merged to form Electric and Musical Industries, or EMI. The new company gave Blumlein a bonus for his work.

By then he was working on binaural stereo for loudspeaker reproduction, not restricted headphone listening. While Bell Labs experimented with lines of loudspeakers, Blumlein used two speakers and the baffle effect of the human head to fool the brain into thinking the sound was coming from a wide spread of different directions. Blumlein's now famous patent UK 394,325, filed in 1931, explains how the system lets the ears register low-frequency phase differences and high-frequency intensity differences.

The original patent text is a model of clarity which contrasts starkly with the incomprehensible rubbish which modern inventors often write, either because they do not understand how their inventions work, cannot explain it, or hope to disguise old ideas with new verbiage.

Blumlein also needed a way of recording both channels from a stereo microphone pair in the single groove of a disc. He did this with the 45/45 system of cutting the different signals on each wall of the groove. In 1933 he made several test recordings of "Walking and Talking." The next year he was allowed into EMI's Abbey Road Studios to cut stereo discs of Ray Noble's dance band and Sir Thomas Beecham conducting the London Philharmonic Orchestra.

FILM, TELEVISION, AND RADAR

In 1935 Blumlein moved onto film, splitting the optical soundtrack of 35mm film into two parallel half tracks to capture stereo. He made a series of test films, including fun stunts and playlets. Decades later, it needed only a small modification of a standard Dolby stereo projector to play back the originals (once they had been transferred from the old, explosive, nitrate stock to safety acetate!). Then, with all things stereo sorted, but no commercial market in sight, Blumlein moved on to television.

In the mid-1930s the British Government was being lobbied by John Logie Baird to adopt his mechanical spinning wheel system, and decided to issue an open challenge: anyone was welcome to try to come up with a better system. EMI and the Marconi Radio Company joined forces, again under Shoenberg, to develop an all-electronic system, which used 405 scanning lines. Blumlein was a key figure in the TV team, patenting vital building blocks such as waveform synchronization by line and frame pulses. But Baird continued to improve his system and he was a great self-publicist.

To settle the matter once and for all, the Government licensed both systems for a trial period towards the end of 1936. Transmissions were broadcast for two hours a day, with the Baird system used one week and Marconi-EMI's the next. In 1937, the Baird system was inevitably rejected, and Marconi-EMI's all-electronic TV became the UK standard. Although it was shut down in the war, the same system re-started and remained working until 1985, after 20 years of parallel running with Europe's new 625 line TV.

By now the situation in Germany was deteriorating, and to some people war seemed inevitable. But in the climate of appeasement, preparing for war was an unpopular policy. The EMI team were quietly moved over to highly secret military research on radar. There is good reason to believe that the British Government encouraged the development of electronic TV as a way of ensuring that the electronics industry would develop high-frequency, high-power amplifier tubes and cathode ray display screens, which would be needed for radar.

Blumlein also worked on highly directional microphones for pinpointing the sound of incoming aircraft.

FATAL ACCIDENT

By 1940 the EMI team had Airborne Interceptor radar working to let British aircraft track German invaders. The next project was H2S, a scanning radar that produced a map of the ground. In June 1942 the EMI team, including Blumlein, was flying in a Halifax bomber to test H2S, with the then-new Magnetron microwave amplifier. The plane crashed, killing all on board, probably because of faulty servicing.

The accident was kept secret, as was Blumlein's death. No one wanted Germany to know what a blow the Allied research project had suffered.

With postwar austerity, hi-fi, stereo recording, and two-channel film remained a very low priority. When the stereo LP standard was set by the RIAA in 1958, Blumlein's 45/45 system was wrongly attributed to Westrex, the Bell Labs subsidiary. In the UK, Percy Wilson, writing in the then-outspoken *Gramophone* magazine, was furious.

The reason for the RIAA's gaffe was simple. No one, outside a small circle of engineers in the UK, had heard of Alan Blumlein or his achievements. A British engineer, Ben Benzimra, set out "to raise this man from the dead," as he put it, and started to collect information for a biography. Benzimra became ill and died, and the job was taken over by a Francis Paul Thomson of Watford.

For literally decades, Thomson collected every available piece of information on Blumlein. He contacted all Blumlein's associates and wrote open letters to technical magazines asking for private papers and personal reminiscences. All those who had known Blumlein and wanted to see him honored jumped at the chance. Initially Thomson had the full support of Blumlein's family, too.

But as time wore on, it became obvious that Thomson had developed an obsession with collecting information (not just on Blumlein but other inventors, too) and was out of his depth on the technology. He had worked briefly at EMI as a lab assistant, but then gone off to write books on banking and tapestry.

As suspicions grew, Thomson became increasing paranoiac over any inquiries about his progress on the Blumlein book. He would reply with legal threats, rambling irrelevancies, questions, and offensive rudeness. He claimed raids on his home trash cans and personal attacks.

By the '80s and '90s Thomson's priority had become to stop anyone else from

to page 62

Book Review

THE CAR STEREO COOKBOOK

Reviewed by Dennis Colin

The Car Stereo Cookbook, by Mark Rumreich. Available as part #BKMH27, for $24.95 from Old Colony Sound Lab, PO Box 876, Peterborough, NH 03458, (603) 924-6371, FAX (603) 924-9467, E-mail custserv@audioXpress.com, 296 pp., shipping wt. 2 lb.

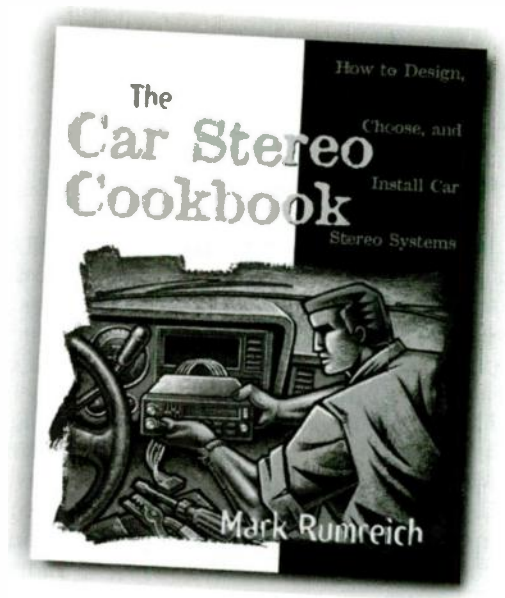
Take it from one who's played with car audio since 1963, this book is both 100% correct and comprehensive. Each subject listed in the table of contents is a well-written and -illustrated discourse which not only describes a spectrum of appropriate systems, but also explains the fundamental electroacoustic principles (such as wavelength relations, bass port function, and Ohm's law).

Chapter 1 ("Before You Begin") covers basic considerations such as available products and options, and solutions to common problems (low volume, muddy bass, and so forth).

Chapter 2 ("Connectors, Supplies, Tools, and Techniques") describes and illustrates a wide variety of connectors and tools, teaches correct soldering, and explains meter use, how to make a speaker polarity tester using a flashlight battery, and many handy tips.

Chapter 3 ("Speakers and Speaker Projects") covers the gamut of available drivers, including surface-mount tweeters, coaxial units, and high-performance "separates" (also called component speakers). The author describes various installations, including doors, dashboards, and kick panels, with their relative pros and cons regarding response smoothness, sonic integration, and imaging. He also describes simple crossovers for add-on tweeters and resistive attenuators.

Chapter 4 ("Subwoofers and Subwoofer Projects") is by far the largest chapter. Starting with configuration options (speaker-level crossovers, bi- and tri-amping, amps with built-in crossovers, crossover types, and amp bridging), this chapter proceeds to the main topic—enclosures and drivers. First, the author discusses the pros and cons of free-air, sealed, ported, bandpass, and transmission line enclosures, and then explains driver Thiele/Small parameters, with emphasis on comparing woofers for sealed, ported, and bandpass applications.



Of particular interest is the author's use of the parameter $f_{ob} = f_s/Q_{ts}$. He says "f_{ob} indicates how low the relative bass response would be in a gigantic box"; it's a means to compare woofer f_3 in a trunk or other approximate infinite baffles ("free air"). Another parameter he defines is $V_{of} = V_{as} \times Q^2_{ts}$.

Fourteen graphs follow, each a family of six curves showing normalized responses for sealed, ported, and bandpass enclosures. These use the parameters f_{ob} and V_{of}. I checked some of the vented alignments against published alignments from a reputable manufacturer, and, well, they aligned. The graphs are sufficiently detailed, wide-range, and accurate to design most any bass enclosure with neither a computer nor a math degree—very nice!

Then a simple formula for determining the ported box resonance frequency follows: $f_b = 0.39\ f_{ob}$. Also shown is the calculation for port length: L = (2117 $D^2 \div f^2{}_b V_b$ – 0.732D, where L = length, D = inside diameter (inches), and V_b = box volume (ft^3).

The chapter also describes in detail multiple driver systems (parallel, push-pull, and isobarik), as well as numerical examples, construction issues, bracing, and damping. Finally, the author addresses system adjustment by ear.

Although this chapter is complete enough to include non-trunk woofer locations, I would have preferred to see more detail about their advantage in clarity (tonal and transient) over the usual trunk placement (See "Real Backseat Bass," *SB* 7/98). But this is no criticism of the book; Mr. Rumreich gives you everything you need to design any specified woofer, so it's up to you to find the best-sounding location within your labor and aesthetic-acceptance budget (as I have been for 36 years). This chapter is equally useful for home woofers, and by itself is worth the price of the book.

Chapter 5 describes head units, covering aftermarket options, format (CD, and so on), functions, wiring, adapters, and installation. The rest of the chapters—entitled "Amplifiers and Amplifier Projects," "Equalizer Projects," "Biamping and Crossovers," "CD Changer Projects," "Accessories," and "Battling Noise"—likewise cover the remaining topics comprehensively, with much detail, illustration, explanation, and an abundance of wiring diagrams.

Overall, this book is a welcome oasis of knowledge and affordable car stereo solutions amidst a desert of info-barren hype in the car stereo magazines ("You need 900W into 3-18″ woofers and the rest of the $10,000 installation!"). And Mr. Rumreich's comprehensive description, explanation, and wide spectrum (simplest through advanced) of solutions will enable you to have first-rate mobile sound with a modest budget.

As an old car-stereo fanatic, I highly recommend this book.

Tools, Tips & Techniques

JUNK TO TREASURE

By Angel Luis Rivera

While driving through Sarasota, FL, I spotted some speaker boxes, ready to be picked up by the garbage man. I pulled into the driveway and glanced at the speakers, which were MCS Series systems from J.C. Penney, but the boxes were in pieces, battered and damaged beyond repair. The 12" woofers' plastic dustcaps, emblazoned with the PYLE logo, were broken. Still the woofer, midrange, and tweeter drivers looked promising.

I wondered whether they worked as I walked up to the front door, knocked, and a man the size of a small building answered. "May I have those speakers if they are being thrown out?" I asked. "Help yourself," he gruffly replied.

TREASURE I

Loading the speakers into my station wagon, I almost immediately regretted taking them, as the chipboard fell apart and crumbled, making a mess. All the way home I chided myself for picking up what was "obviously" hopeless garbage.

When I got home, I took the boxes apart and threw out the "wood." I unfastened the drivers and hardware, amazed at the quality of the drivers and the midrange and tweeter level control switches. I was also pleased with the inductors used throughout the crossover networks; they appeared to be of good quality and some were even air-core coils. Unfortunately, the magnets had been knocked off one of the midranges and both of the tweeters, while the magnet on the midrange dome was fairly large, and the whole unit looked well built.

But the shocker was the woofer: a huge magnet with a vented pole piece! Closer inspection showed the cone was made of heavy coated paper with a rubbery-foam-like surround. The whole woofer reeked of quality construction.

For those of you contemplating running out and buying these systems just to get your hands on the PYLE woofer, I seriously doubt that these were the original woofers, judging from the shape of the boxes when I found them. Most likely the owner damaged the original woofers and replaced them with the PYLE drivers. I tested them with an AA battery and, lo and behold, they worked!

The dustcaps, mentioned previously, were crushed and broken, so I removed them and tried repairing them with glue, but to no avail. I realized all I needed was something light, plastic, rigid, airtight, and inexpensive to use in place of the dustcaps. I found the perfect material at SCOTTY'S for 89 cents—a plastic yard sale sign.

I measured the diameter of the dustcaps and used my compass to draw out two circles on the yard sale sign. I cut them out with scissors, glued them in place, and voilà! My new-found woofers now give evidence of their pedigree by

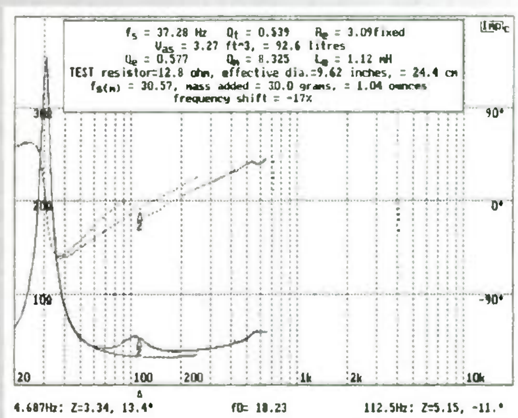


FIGURE 1: IMP graph of YA speaker.

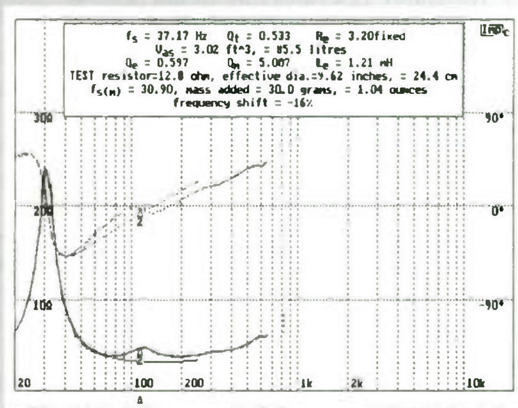


FIGURE 2: IMP graph of RD speaker.

STATEMENT OF OWNERSHIP, MANAGEMENT AND CIRCULATION

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I certify that the statements made by me above are correct and complete. Publication number 529-310.

Edward T. Dell, Jr., Publisher

their newly acquired bright yellow-lettered dustcaps: one reads "YA," the other "RD" (*Photo 1*). And they work beautifully (*Figs. 1* and *2*)!

TREASURE II

As luck would have it, the very next day I was driving with my wife (in her car this time) close to home and spied a gentleman cleaning out his garage: several speaker boxes were at the front of the pile. I hung a quick U-turn, pulled up alongside the pile of throwouts, and squinted in the direction of the boxes. "What does the logo say?" I asked my wife. "Oh, I don't know," she replied. "It's spelled K-E-F."

I needed to hear no more! I jumped out of the car and spoke to the owner, who told me neither speaker was working but I was welcome to them. I drove home and a few minutes later I returned with my handy little station wagon.

This time my find was the KEF Carina II—two 7.5″ woofers flanking a 1″ dome tweeter in a D'Appolito configuration. I'd hit the jackpot! The boxes were in fair shape, although one of them had become wet. The particleboard thickness was a skimpy ½″ for such fairly large boxes (23⅝″H × 11⅞″W × 9½″D). They couldn't pass the knuckle-rap test, being pretty resonant.

I removed the drivers. The woofers had nice rubber surrounds with plasticized paper woofer material, moderate-size magnets, and stamped metal frames. I tested each driver with a battery and music, and to my delight found that they all worked. The tweeters were quite sweet sounding.

I turned my attention to the crossovers (*Fig. 3*). The tweeter filter was a third-order circuit with one of its capacitors blown up like a firecracker. The woofers were wired in series with a Zobel circuit parallel to them and a coil and capacitor in series.

The capacitor values were easy to deduce but not so with the resistor and two inductors—I would need to remove these from the circuit to measure their values and it wasn't worth the effort, so they shall remain a mystery. Polyester fiberfill had actually melted and hardened into a glob of plastic around the first capacitor and a coil, suggesting great heat had passed through there. Holy smokes (pun intended), I'm surprised it hadn't caught on fire.

MODIFICATIONS

My challenge now was to restore these speakers to working condition. But I wished to do more than that—I preferred to modify them and demonstrate what is possible for the home hobbyist to accomplish, with as little expenditure as possible, some investment of time, effort, and a bit of resourcefulness.

I replaced the blown electrolytic capacitor with Mylar caps and bypassed the other electrolytic caps with small-value Mylar and polyester capacitors I had on hand. I then hooked up the crossovers to their respective drivers and to Bill Waslo's IMP and plotted their responses (*Figs. 4* and *5*). They were identical in response and appeared undamaged—apparently that first blown capacitor served as a "fuse" that "blew" and protected the rest of the crossover components downstream.

I left the modi-

PHOTO 1: The PYLE drivers.

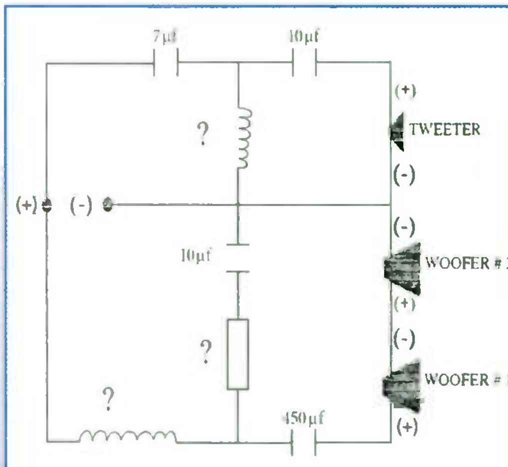

FIGURE 3: KEF crossover.

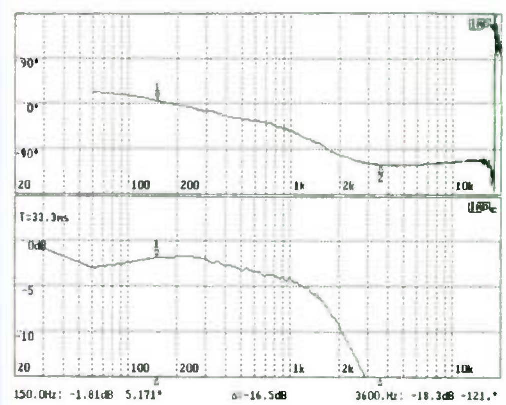

FIGURE 4: Crossover response of KEF drivers (low-frequency).

PHOTO 2: Crossover (left outside the enclosure) connected to speaker posts.

PHOTO 3: The refurbished KEF speaker.

Book Review

Continued from page 58

writing a biography of Blumlein, even though he had no hope of producing one himself. In the last years of his life (he died earlier this year), Thomson threatened to burn all the material he had collected. It is still unclear what has become of the vast collection. No one has ever had any way of knowing what he collected, anyway.

THE FINISHED PIECE

So how was Robert Alexander able to write a book? After an initial tangle with Thomson, he gave up even trying to access the material which the self-styled biographer had squirrelled away. Instead, Alexander sought the assistance of EMI's archivists, read everything that Blumlein had written in his patents, and even arranged for them to be retyped and posted on the Internet (www.jayjaybee.com). He also went to a string of libraries and museums, including the Imperial War Museum in London, the Newspaper Library, the Patent Office records, the Royal Airforce Museum, and the British Library of Recorded Sound. Over a period of five years, and despite Thomson's solid obstruction, Alexander managed to put together a definitive collection of all available factual information.

Focal Press, a publisher of technical books, had never previously handled a biography, but made an exception. The result is a definitive biography which provides long overdue documentation of Blumlein's life. Engineers, unless they are nit-pickers, will surely welcome it. So will anyone with an interest in audio and electronics history. But this book is not an easy read and is unlikely to spark mass market interest.

Personally, I have always believed it would be impossible for anyone to write one book that tells the human story of Alan Blumlein, while at the same time doing justice to his engineering achievements. That is why there are around 60 biographies of Thomas Edison. Reading this book does not change my mind. But Robert Alexander has done what no one else has done, and I salute him for it.

fied crossovers outside the enclosures (*Photo 2*) for ease of access, further experimentation, and possible additional modification later on, but also to facilitate future multiamping. Multiamping, in turn, will confer additional flexibility, allowing me to wire the woofers in parallel for greater efficiency, if I so desire. Nor would I be limited to these woofers and tweeters if I can find superior drivers that would work well in these particular enclosures.

I have a GSI two-way second-order electronic crossover that crosses over at 2kHz. My friend, Matt Hamilton, and I will be experimenting with this unit in conjunction with the passive networks to see what effect this has on response and sound quality when the two are, in effect, cascaded together.

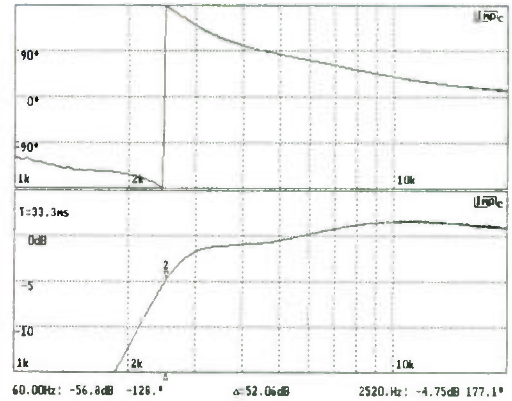

FIGURE 5: Crossover response of KEF drivers (high-frequency).

Each woofer and tweeter is connected to its own set of dedicated speaker posts (*Photo 2*) (six posts per side) for ease of connection and adaptability, as referred to previously. I even replaced the original speaker binding posts' washers, which appeared to be made of some anodized metal, with stainless-steel types to avoid possible rectification effects.

I used the original enclosures as endoskeletons and added ¾″ particleboard around all six sides of the boxes, making each top, bottom, side, front, and rear baffle about 1.25″ thick. The original external box size did not include a ¼″ lip around the front and rear baffles, which I sanded down so both front and rear baffles were then flat. The new external dimensions are 25-⅛″H × 13-⅛″W × 10¾″D after modification. Corners are rounded to minimize diffraction effects (*Photo 3*).

FINAL TOUCHES

Matt stated that because particleboard is porous and breathes, the enclosures should be sealed on the inside. I used primer sealant to accomplish this. I sanded the exteriors with an orbital sander until that particleboard was as smooth as glass. I applied a lacquer-based wood-sealant that dried quickly (in one hour), which allowed successive coats to be applied within relatively short periods of time.

I finished them with Formby's Tung Oil (oil/polyurethane), which couldn't be easier to apply: put on a rubber glove to protect your skin, pour the Tung Oil onto a cotton rag (for example, a T-shirt or pajamas) and spread it on; it glides effortlessly and goes on smoothly. I must say that for particleboard boxes, they sure look beautiful. In addition, the new boxes are heavier, denser, more rigid, airtight, better braced, and consequently far less resonant.

I filled and covered the back of the stamped metal woofer frames (which were thin and "hollow") with silicone caulk. The original unmodified frames rang like a bell when struck by a fingernail or plastic or metal tool—now they are nicely damped and are relatively dead. They produce a "thunk" when struck and do not ring. The silicone also serves double duty as a gasket to decouple the woofers from the cabinet, as well as working as a sealant for an airtight fit. I laid the stock drivers flat on top of the original baffle.

I figured I'd go KEF one better and recessed the speaker drivers in their new cabinets so that they are now flush with the baffle front. Actually, Matt and I miscalculated and routed out a larger area for the woofers than called for, but I used this to advantage and siliconed around the woofers to further decouple them from the box. These maneuvers should help cut down on resonances even more and further improve imaging and sound quality. It also gives the speakers a distinctive look.

ACKNOWLEDGMENTS

Many thanks to my friends Matt Hamilton and Dave Currier for their help and advice with routing, sanding, and finishing the speaker cabinets. Particular thanks to Matt for his invaluable help in measuring the drivers and their crossover networks; and special thanks to Dave for all he's taught me about wood and use of his woodshop and tools.

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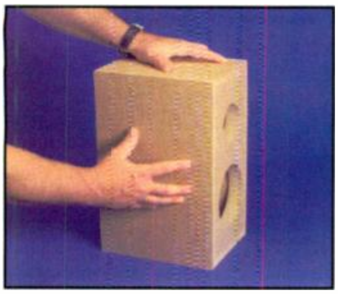
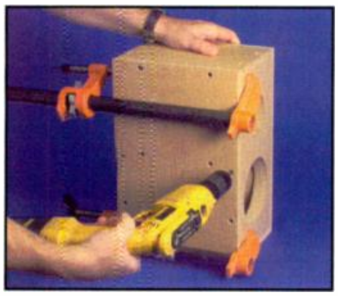
By using technology previously only available to high end speaker manufacturers, we are able to offer audiophile speaker cabinets at reasonable prices. Now you can build an audiophile quality speaker system without expensive woodworking tools or hours of labor. We have developed a series of cabinets designed by audiophiles and computer manufactured. The cabinets are made of 1" MDF (medium density fiberboard) to reduce cabinet resonance and coloration. They are precision milled with a CNC router to incredible exacting tolerances. Dado and slot joinery are used throughout to provide incredible strength. Internal braces reduce unwanted cabinet resonances. All you have to do is assemble, glue, and finish to your liking.

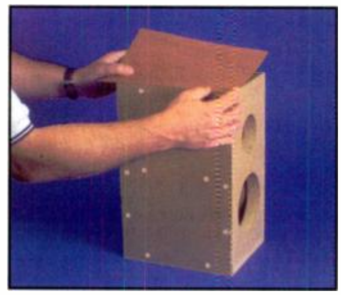
Blank Baffles: The cabinets come with pre-cut baffles for standard drivers. Blank baffles are available for you to rout yourself for flush mounting or custom drivers.



Step 1
Assemble cabinet on your workbench and pre-drill and countersink screw holes. Modify driver holes if necessary at this time.



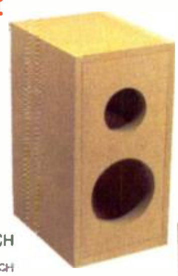
Step 2
Brush Polyurethane glue (available at most home centers) on both sides of all joints. Assemble cabinet (clamp if possible) and install screws.



Step 3
Fill screw holes with putty let dry and sand. Apply your veneer or laminate.

Step 4
Varnish or paint cabinet to your liking. Install crossover, rear terminals and drivers.

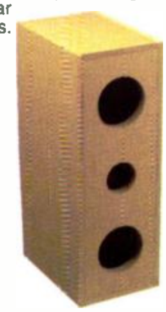
.25 Cu. Ft. Bookshelf Cabinet



Reminiscent of the classic British mini-monitor, this cabinet is well suited as a compact 2-way audiophile system or home theatre setup. .25 cu. ft. internal volume. Includes 1" Dado brace. 1" Front Baffle is pre- cut to accept most 5 - 6 in. mid/woofs and tweeters. 100% 1" MDF Construction. Internal dimensions: 5" W x 12-3/4" H x 7" D ◆External: 7" W x 13-3/4" H x 9" D ◆Woofer hole: 4-5/8" ◆Tweeter hole: 3" ◆Net weight: 12 lbs.

#300-704 .... $34.50 EACH
#300-703 (Blank Baffle, 1" MDF) .... **$3.95** EACH

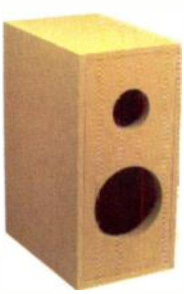
.75 Cu. Ft. Dual Woofer Cabinet



Build the popular D'Appolito style dual woofer speaker system. The front baffle is pre-cut to accept two 5-6" mid/woofs and one tweeter. Popular tall European style cabinet with two internal braces to help eliminate unwanted panel resonance. .75 cu. ft. internal volume including 1" braces. Internal dimensions: 6-1/4" W x 21-3/4" H x 10" D ◆External dimensions: 8-1/4" W x 23-3/4" H x 12" D ◆Woofer holes: 4-5/8" ◆Tweeter hole: 3" ◆Net weight: 26 lbs.

#300-714 .... $59.80 EACH
#300-713 (Blank Baffle, 1" MDF) .... **$7.95** EACH

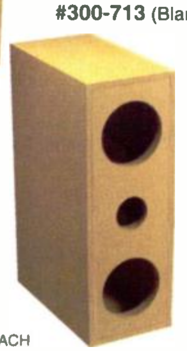
.55 Cu. Ft. Cabinet



This is the perfect cabinet for any single 6-1/2" woofer and 5-1/4" single or dual woofer arrangement. .55 cu. ft. internal volume including 1" MDF Dado brace. 1" front baffle pre-cut to accept 6-1/2" woofer and tweeter. Internal dimensions: 6-1/4" W x 15-1/2" H x 10" D ◆External dimensions: 8-1/4" W x 17-1/2" H x 12" D ◆Woofer hole: 5-5/8" ◆Tweeter hole: 3" ◆Net weight: 20 lbs.

#300-708 .... $46.80 EACH
#300-707 (Blank Baffle, 1" MDF) .... **$5.95** EACH

1 Cu. Ft. Esoteric Speaker Cabinet

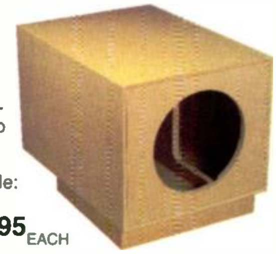


When completed, this cabinet will rival audiophile systems costing thousands! Designed to accept dual 6-1/2" mid/woofs and a center tweeter in a D'Appolito configuration, or buy the optional blank front baffle to design your own system. Two internal braces to help eliminate unwanted cabinet resonance. Internal dimensions: 6-1/4" W x 21-3/4" H x 13-3/4" D ◆External dimensions: 8-1/4" W x 23-3/4" H x 15-3/4" D ◆Woofer holes: 5-5/8" ◆Tweeter holes: 3" ◆Net weight: 31 lbs.

#300-718 .... $69.50 EACH
#300-713 (Blank Baffle, 1" MDF) .... **$7.95** EACH

Note: All of our MDF cabinets are shipped "knocked down" ready to assemble. They include detailed assembly instructions and finishing recommendations.

3 Cu. Ft. Subwoofer Cabinet



Finally, a high quality, high performance MDF subwoofer cabinet at an affordable price. This cabinet utilizes 3/4" MDF (medium density fiberboard) not particle board. MDF has far superior sound deadening characteristics than particle board, plywood, or OSB. 3 cu. ft. cabinet is perfect for 10"-15" subwoofers. Internal Dado brace to reduce cabinet resonance. Inside dimensions: 16-1/2" W x 14-1/4" H x 22-1/2" D ◆Outside dimensions: 18" W x 15-3/4" H x 24" D ◆Woofer hole: 11-1/8" ◆Net weight: 43 lbs.

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Through the course of the years Morel brought many unique and innovative products to the speaker industry. The introduction of the 3″ voice coil in a 5″ basket, using hexagonal shaped aluminum wire, utilizing a double magnet system and ducted design woofers and mid-basses are a few examples of the company's breakthroughs. Also introduced were the Integra concept (single motor system for both the tweeter and woofer) and the Push-Pull 8″ and 10″ subwoofers (dual motor system, dual voice coils with a single cone).

Integra

Push-Pull

Double Magnet

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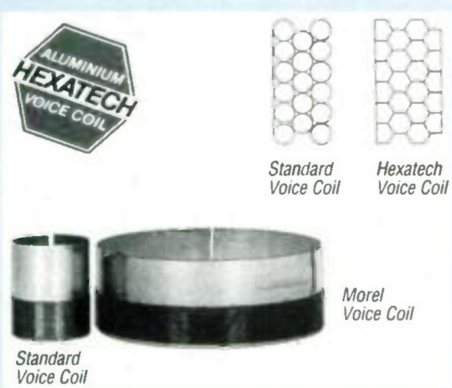

Typical double magnet ducted woofer.

Double magnet tweeter

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