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Speaker Builder

THE LOUDSPEAKER JOURNAL

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Koonce & Wright

BUILDING THE LOW-COST 3-WAY MIRACLE SPEAKER

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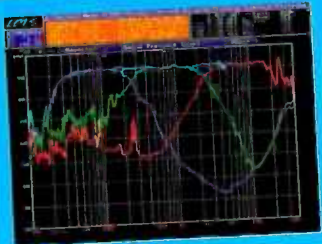
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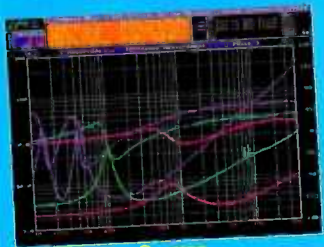
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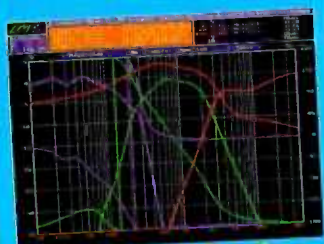
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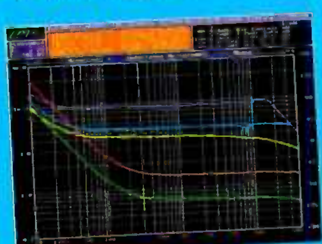
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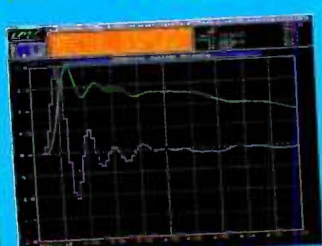
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Reader Service #101

■ ON-SITE SPEAKER DESIGN

True Image Audio invites you to visit its site on the World Wide Web to obtain new product information, special promotions, and a link to other areas of interest to the audiophile. The company recently reduced the price of its Speaker Design Toolbox, available as MacSpeakerz for use on a Macintosh computer or as WinSpeakerz for Windows 95 systems.

A demo version of the loudspeaker-design application software may be downloaded from the website.

True Image Audio, 349 W. Felicita Ave., Ste. 122, Escondido, CA 92025, (800) 621-4411, FAX (619) 480-8961, http://members.aol.com/spkrtools.

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■ CLASSIC COMEBACK

Now 15-years old, the Quad ESL-63 returns to the US market in high-end audio specialty stores. This electrostatic loudspeaker's floor-standing design measures 36" high by 26" wide, accented by a curved frame, transparent black grille, burnished wood end pieces, and matte-black pedestal. Radiating as a dipole and mounted in an open frame, the ESL-63's full-range diaphragm is one-tenth the thickness of a human hair, coated with conductive material, negatively charged, and located between two perforated metal stators for maximum responsiveness. Distributed by Mission Electronics Inc., 400 Matheson Blvd. East, Unit 31, Mississauga, ON I4Z 1N8, Canada, (800) 838-7955, FAX (905) 507-0797.

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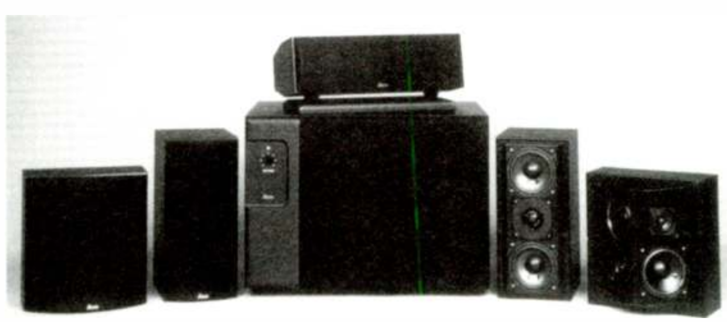

∩ DOLBY UPGRADE

Atlantic Technology has upgraded the home-theater System 250.1 to meet the greater power, bass, and surround demands of Dolby Digital. The system incorporates three speakers: model 251.1 LR front-channel, model 253.1 C center-channel, and model 254.1 SR full-range surround. Models 251.1 LR and 253.1 C each have a 4" mid-woofer with a long-throw voice coil, ¾"-thick cabinet walls, and heavy-duty crossovers. The 254.1 SR surround speaker allows the localization of rear-channel sounds and boasts a frequency response of 80–20kHz at ±3dB, a front baffle with two ½" tweeters and two 4" woofers, and phase-inverted twin-pole driver configuration. Atlantic Technology International, 343 Vanderbilt Ave., Norwood, MA 02062, (617) 762-6300.

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About This Issue

Speaker enclosures come in all shapes and sizes, but have you ever mounted a driver in a barrel? **Charles T. Pike** takes this unusual approach and explains how the advantages outweigh the disadvantages in "9Hz in a Barrel" (p. 8). This project, which uses a 55-gallon polypropylene drum, features one of the easier-to-build enclosures we've seen in a while. And the results are enjoyable low-frequency responses you can live with.

When you design loudspeakers, phase and signal delay are important to consider, but few of us are really comfortable with these concepts. In answer to reader questions, **Bill Waslo** provides a short tutorial entitled "Time, Frequency, Phase, and Delay" (p. 12) and defines these phase-response terms.

Now that we've selected the drivers and design for our three-way system ("A Modest-Cost Three-Way Speaker System, p. 20), it's time to begin construction. In part 2, **G.R. Koonce** and **R.O. Wright** present detailed illustrations, with plenty of construction and assembly tips, to guide you through the process and ensure good-sounding, good-looking finished boxes.

Family togetherness extends into the speaker-building world as the father and son team of **Richard** and **Erin Honeycutt** offers guidelines on how to choose inductors for crossover networks ("Inductors for Crossover Networks," p. 36). You'll discover that their advice on choosing components for passive crossovers debunks many of the traditional beliefs we've held about inductors.

Also in this issue, we've devoted plenty of space to answering reader questions and comments (*SB* Letters, p. 50, and Ask *SB*, p. 46). And be sure to check out "Tools, Tips and Techniques" (p. 62) for a simple modification that improves the clarity of a Vifa tweeter.

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JOHN STUART MILL

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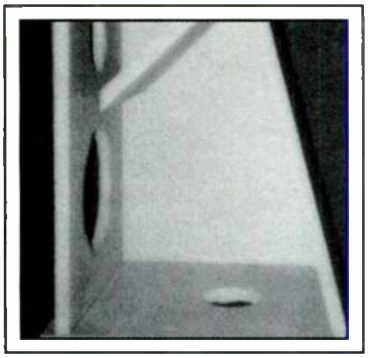
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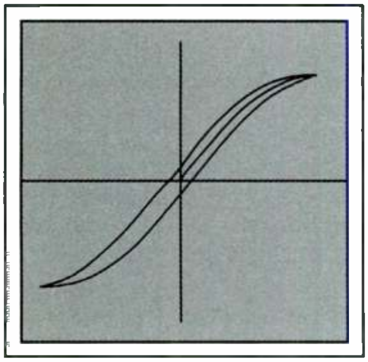
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9HZ IN A BARREL

By Charles T. Pike

This article describes an easy-to-build subwoofer that has a low-frequency cutoff of 9Hz. After completing my previous article ("A 16Hz Subwoofer," *SB* 4/94, p. 26), I had a Radio Shack 15″ Model 40-1301 woofer without an enclosure, and since I find nothing more inspiring than having a driver without a home, I started thinking about enclosure designs. I did not want to build a copy of my previous design because it takes up a lot of space and it's more fun trying something new.

The first approach I considered was mounting the speaker in the wall of my family room with the back open to a 2000ft^3 storage area under my living room. *Figure 1* is a printout from the BassBox 5.1 computer program showing the calculated sound pressure level (SPL) as a function of frequency (top), maximum acoustic power (center), and impedance (bottom) for the Radio Shack woofer mounted in a 2000ft^3 infinite-baffle enclosure. Although the frequency response could be corrected, as pointed out in my previous article, the maximum SPL is limited at frequencies below 50Hz by the driver cone excursion.

OTHER APPROACHES

Since this approach did not seem too promising, I decided to look at alternatives. A fourth-order bandpass enclosure is normally divided into two sections, with a sealed volume in back of the speaker and a ported volume in front. All of the output then comes from the port. In examining bandpass enclosures, it occurred to me that if you were to replace the rear sealed box on a fourth-order bandpass enclosure with an essentially infinite baffle, it should be possible to extend the low-frequency response.

The top graph in *Fig. 2* shows the calculated frequency response of such a system with a bandpass response that extends from 8.9–45Hz at the 3dB down points! In addition, the center graph in *Fig. 2* shows that the maximum output is not limited by speaker excursion for frequencies above about 16Hz.

Of course, there is a price to pay for this good low-frequency response: a loss in efficiency. As shown in *Fig. 2* (top) the maximum SPL is 83dB at 2.83V RMS input. However, at 100W, the maximum input power of the speaker, the output would be about 103dB, which I think is more than adequate.

CONSTRUCTION

You can calculate the optimum volume for the ported section of the enclosure with the formulas given by Jean Margerand ("The Third Dimension: Symmetrically Loaded," *SB* 6/88, p. 29) or by using the Quick Box computer program. For this design, the volume is about 210 ltr. To simplify construction (and since it would be out of sight under the house), I decided to use a 55-gallon drum—almost exactly the correct volume.

I chose a 55-gallon polyethylene insecticide tank because it is easier to cut than a steel drum. On one end of the tank you must provide a flange for mounting the speaker. Cut a 16″ square out of ¾″ plywood or particleboard, with a 13 5/8″ hole for the speaker. Bolt it to the tank with 8 × 10–24 flat-head bolts countersunk to provide a smooth

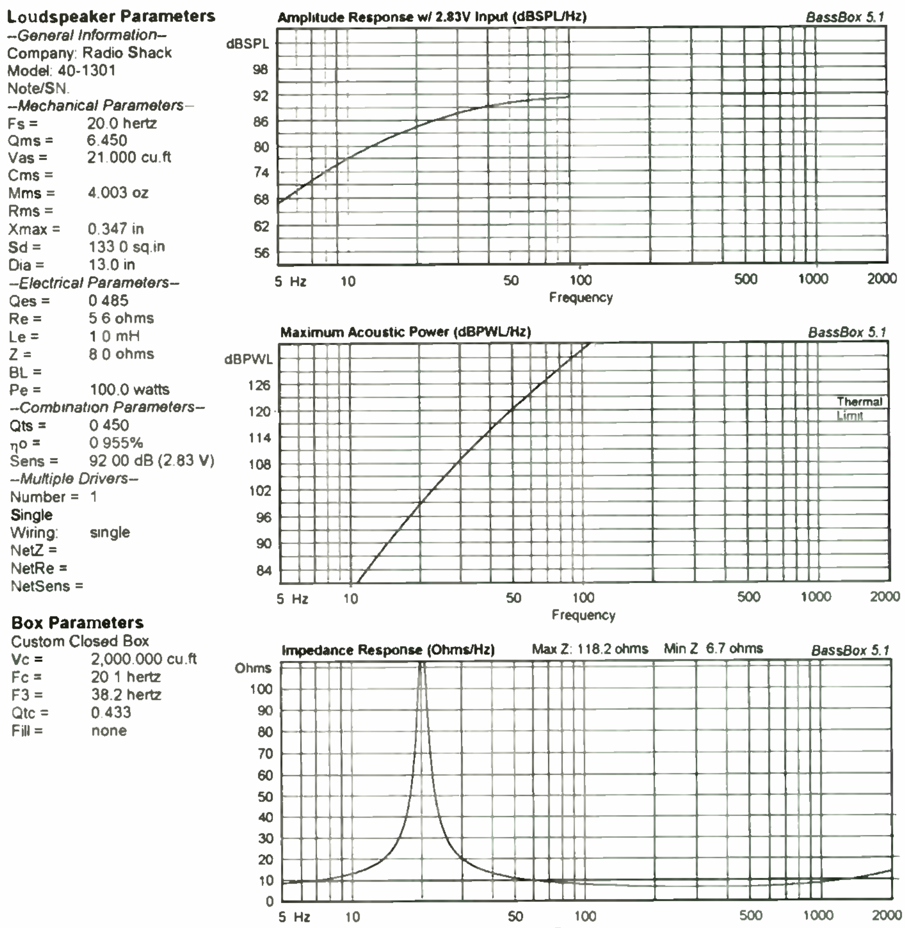

FIGURE 1: Calculated speaker performance in infinite baffle.

ABOUT THE AUTHOR

Charles Pike, a long-time audio enthusiast and music lover, designs and builds his own loudspeakers and amplifiers. He is primarily interested in music of the baroque and classical periods. He has an MS in Physics from Fairleigh Dickinson University and is professionally occupied in designing lasers for military applications.

mounting surface for the speaker.

If you use plywood for the flange, make sure to fill with wood putty any voids that might make air leaks. Use floor-tile cement between the flange and the drum to make a tight seal. *Photo 1* shows the flange installed on the drum and *Photo 2* the flange with the speaker mounted.

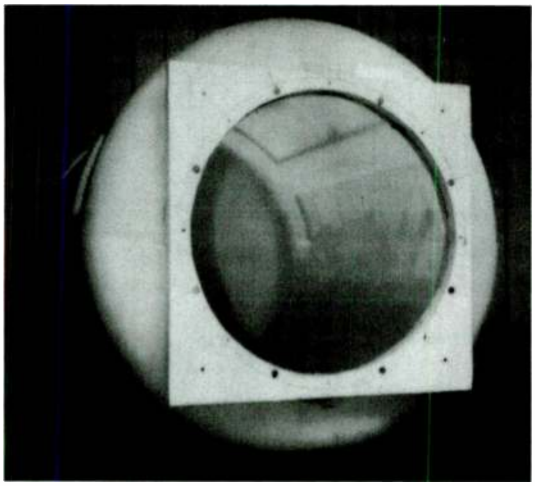

PHOTO 1: Speaker mounting flange.

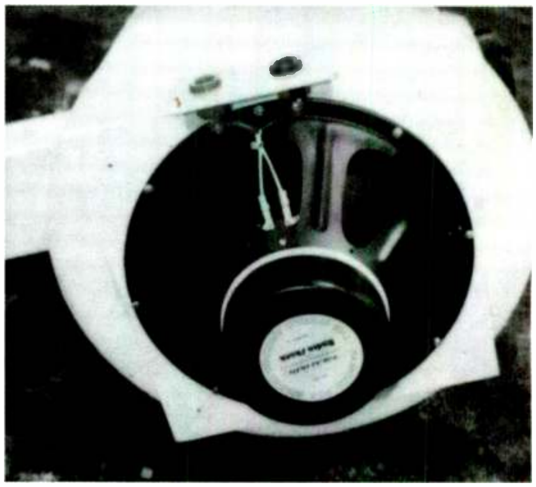

PHOTO 2: Speaker mounted on flange.

Using ¾″ plywood or particleboard, make the port duct 17″ long, with inside dimensions of 5″ × 5¼″. Provide a small flange for bolting the port duct to the drum wall. Cut a 5″ × 5¼″ hole in the rectangular protrusion where the drain is located in the bottom of the tank, and attach the duct using screws and floor-tile cement for an airtight seal. *Photo 3* shows how the duct is fastened to the drum.

Then stiffen the drum with two lengths of perforated plumbing strapping from your local hardware store, tensioned with small turn-buckles. Plug the small hole in the cover with a screw or silicone sealant. *Photo 4* shows the completed system, and *Photo 5* shows it installed under my living room.

TESTING

In using a drum for the enclosure, I was concerned as to whether it would be rigid enough. It does vibrate somewhat in use, but the frequency response (*Fig. 3*) measured with a Panasonic P9932 microphone (which J. D'Appolito indicated is flat to 10Hz [letter in *SB* 3/94, p. 66]) agrees within 1dB with the computed curve.

Figure 4 shows the impedance, which also agrees well with the computed curve. It is interesting that fourth-order bandpass enclosures seem to follow the old rule of thumb that the port should be adjusted to make the height of the two impedance peaks approximately equal.

I measured the near-field SPL at the port output to be 111dB at 2.83V RMS input. D. B. Keele[1] gives the following relationship for calculating the near-field sound

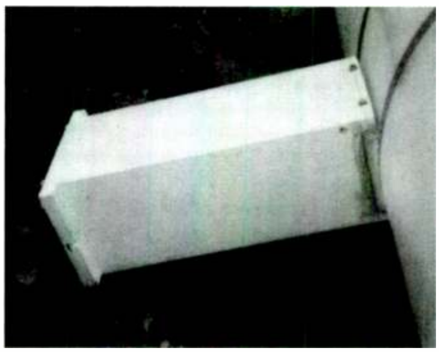

PHOTO 3: Duct mounting.

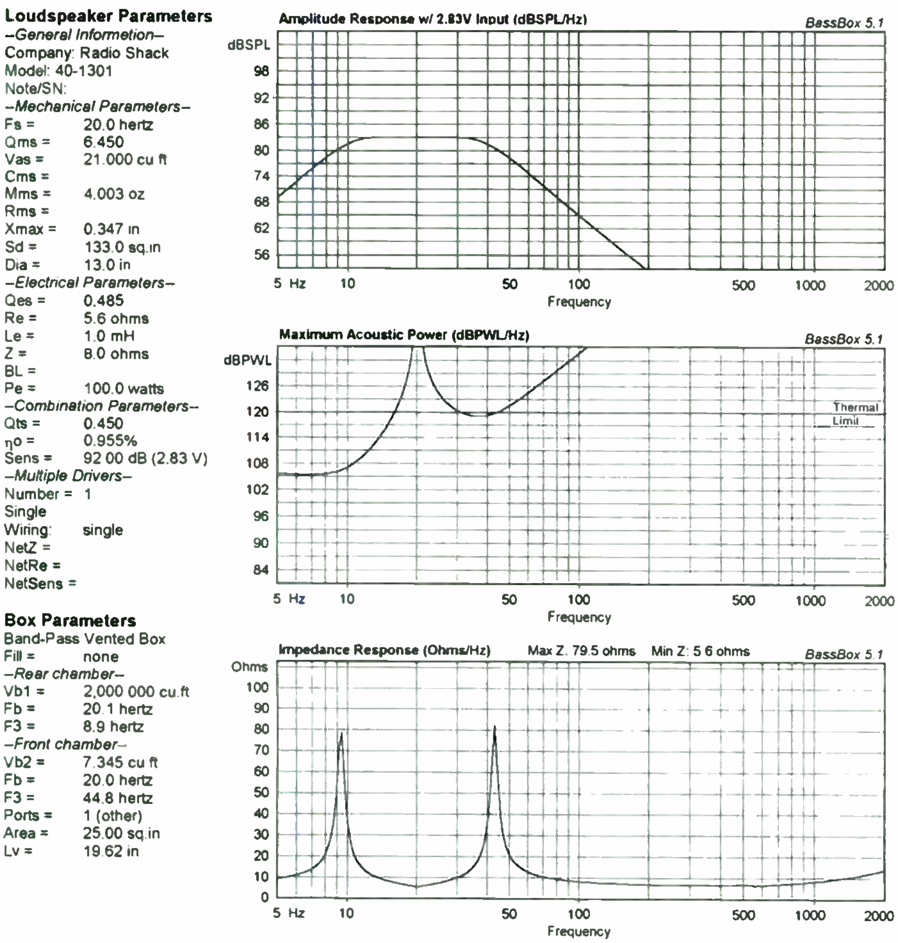

FIGURE 2: Calculated speaker performance in modified fourth-order bandpass enclosure.

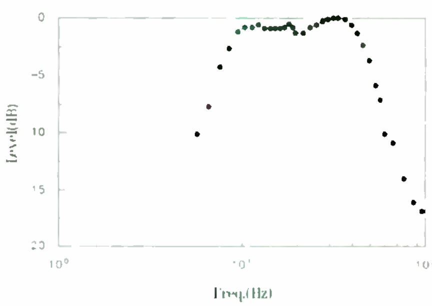

FIGURE 3: Measured frequency response.

pressure from the far-field value for a source radiating into a hemispherical space: Pn = Pf(2r/a), where Pn is the near-field sound pressure, Pf the far-field pressure, r the distance from the source to the measurement point, and a the radius of the source.

Dividing by the 0dB reference pressure, then taking the logarithm of both sides and rearranging terms, we get: SPL(f) = SPL(n) −20log(2r/a), where SPL(f) is the far-field sound pressure level and SPL(n) the near-field. Take a as the radius of a circle with the same cross-section area as the port. Then, if you let r = 1m, the calculated SPL at 1m with 2.83V RMS input is 82dB, which agrees very well with the 83dB given by the computer model. The testing seemed to indicate that the system performed almost exactly as expected.

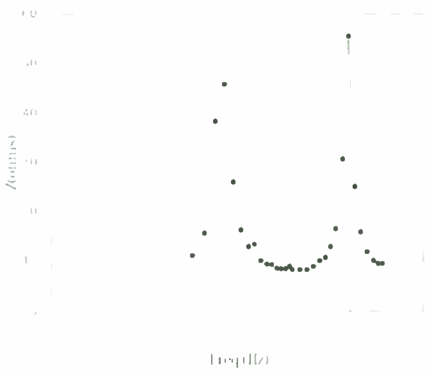

FIGURE 4: Measured impedance.

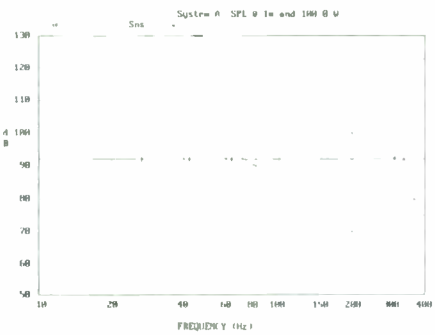

FIGURE 5: Frequency response of system calculated with Bandpass Boxmodel.

CROSSOVER

Figure 5 is the computed response of the system using Bandpass Boxmodel, showing it predicts a port resonance at about 300Hz. Some sort of a crossover is therefore required to suppress this peak. I used the same circuit given in my previous article (*SB* 4/94), with the 0.1µF capacitors—which determine the crossover frequency—changed to 0.033µF. This rolls off all frequencies above 60Hz at an 18dB/octave rate.

LISTENING TESTS

This subwoofer is connected to the right channel of my system, and a modified version of my 16Hz subwoofer is connected to the left. They are driven by an 80W-per-channel amplifier that I designed and built.

Recordings such as Jean Guillou's performances of Bach's Toccata and Fugue in D Minor on Dorian DOR-90134 and Mussorgsky's Pictures at an Exhibition on Dorian DOR-90117 are truly thrilling to lis-

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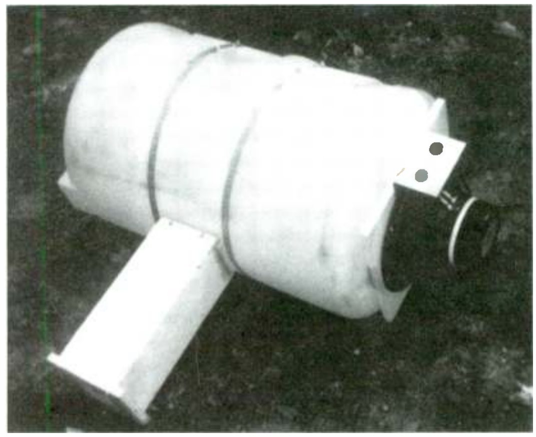

PHOTO 4: Completed system.

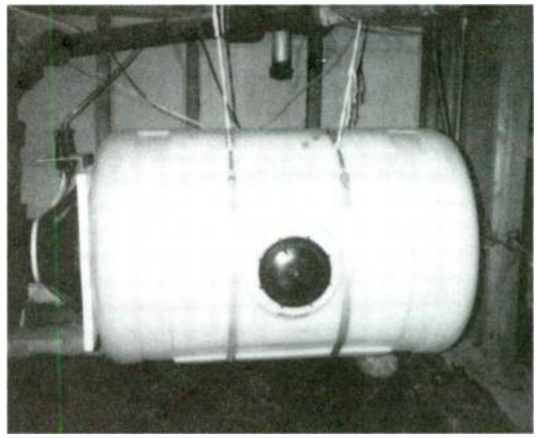

PHOTO 5: System installed.

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ACCESS 6B	Surprising weight in the low end, which is powerful and well defined. This little woofer does not seem to be much embarrassed by any signal.
ACCESS 7A	Extremely smooth 7" midrange/midbass. Excellent prospect for a simple efficient 2-way bass reflex , It can also be used in surround systems main speakers, with or without a subwoofer.
ACCESS 7DB	Outstanding 7" dual voice coil midbass. Powerful, crisp and efficient, this drive unit will play anything your amplifier will throw with panache and relentless enthusiasm.
ACCESS 8A	Very efficient (93 dB +) 8" midrange/midbass. One of the truly rare 8" that can be used in 2-way designs. You will need a very good and efficient tweeter to match this unit (at least 92 dB/W/m)
ACCESS 8DB	Outstanding 8" dual voice coil midbass. Efficient, smooth and crisp sounding it is also capable of handling large dynamics and true low frequencies with the authority of a much larger woofer.
ACCESS 10A	Impact & dynamics (94 dB/W/m). If a 10" midbass can make it to the tweeter range, this is the one. Its nearly perfect roll-off will allow direct wiring without filtering. Rare unit for 2-way 10" designs !
ACCESS 10B	Deep and musical 10" woofer. Ideal for medium size 3-way systems , with one of the 4", 5" or 6" ACCESS midranges. A classic.

TIME, FREQUENCY, PHASE, AND DELAY

By Bill Waslo

Signal theory is a topic that intimidates many audiophiles, who generally have a good grasp of the concept of time and feel comfortable discussing frequency, but consider phase and signal delay to be magical quantities in some sort of esoteric alchemy, understandable only with great effort and through extensive mathematical abstraction. In this article I will attempt to present a relatively simple treatment of these subjects, with emphasis on their importance in loudspeaker design and testing.

MEANINGS

First, I want to clarify the meaning of time-domain data and frequencies. Time-domain data is any quantity that varies with time, which in the case of loudspeakers is usually a sound-pressure or signal voltage. Music, as viewed on an oscilloscope (*Fig. 1*), is an example, as are the various other signals (impulses, MLSs, sweeps, tone bursts, and noise) used in loudspeaker tests.

You might think that frequency is another word for pitch, describing the notes of a musical scale, but for this discussion, frequency is a parameter relating only to sine-wave-like signals. The output of very few musical instruments approximates a sine-wave shape. While you may hear a pitch when listening to an acoustical sinusoid (a sine-wave-like waveform), a note from an instrument such as a piano is not a single sinusoid but a combination of many. In most cases, the note is identified with the lowest strong sine-wave frequency present, but in some cases you can hear a pitch corresponding to a low frequency that isn't even there to any appreciable degree.

WAIT TILL THIS SINE WAVE IS FINISHED

The sine wave upon which this concept of frequency is based is a special repeating time-domain signal. An ideal sine wave is forever; it has no beginning or end. It (along with its phase-shifted alter-ego, the cosine wave) has a specific value at any given time.

To minimize the confusion, I'll refer to points in time using numbers to indicate positions. "Time = 0" doesn't mean "the beginning of time," but just some convenient reference point, with all previous time denoted in negative seconds and all later time in positive seconds.

Mathematical sine waves and cosine waves have a maximum size (amplitude) of 1. The frequency of these waves is a measure of how many times per second the basic sine-wave shape repeats itself. If the waveform cycles only a finite number of times and then stops, it is not a sine wave.

Cosine waves and sine waves are identical except for a phase or time shift of 90°, which means 90/360 (or one-fourth) of the basic waveform. At time = 0, sine waves have a value of 0 and cosine waves a value of +1. Cosines rather than sines are generally used in discussions of Fourier theory.

FIVE-MINUTE FOURIER THEORY

J.B.J. Fourier (1768–1830) was a mathematician and physicist who showed that any time-domain signal can in principle be made from a sum of sized and delayed cosine waves. This means that you could, at least in theory, put a great number of cosine waves through a network that adds the voltages (shifted and amplified) together at each point in time, and duplicate any possible time-domain signal at the output.

Each individual cosine wave varies over time between positive and negative values, and the summation at any point can therefore be positive, negative, or zero (*Fig. 2*). If you are uncomfortable with the idea that your treasured recording of Beethoven's Ninth Symphony could be duplicated with just a bunch of sine-wave generators, I should mention that it would require an infinite number of them; this is an abstract theory, you see.

The Fourier transform, used in many signal analyzers, essentially breaks a time-domain waveform into its component cosine waves. The transform does this by revealing the size and phase position required of the cosines at each frequency to reconstruct the original waveform.

As an example of Fourier summation, a square wave contains cosine waves (delayed and sized) only at the frequencies that are odd multiples of the repetition rate. *Figure 3* shows a limited selection of the first five odd harmonics summing up to make a fair approximation of a square wave.

The classic nonrepeating waveform is the impulse, which has a value of 0 everywhere except at time 0, when it is at plus infinity. These idealized spikes don't happen in real life, of course, yet they are central to the concept of frequency response. If you could take all the cosine waves at all possible frequencies, all with the same amplitude of 1 and all defined at the same time = 0, and play them simultaneously, they would sum to form this impulse.

Another way of stating this is that the

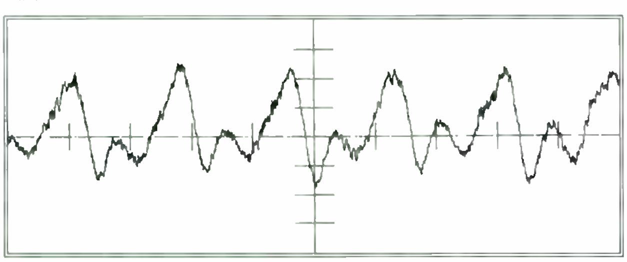


FIGURE 1: A time-domain plot of a sound signal. X-axis is time, Y-axis is sound pressure.

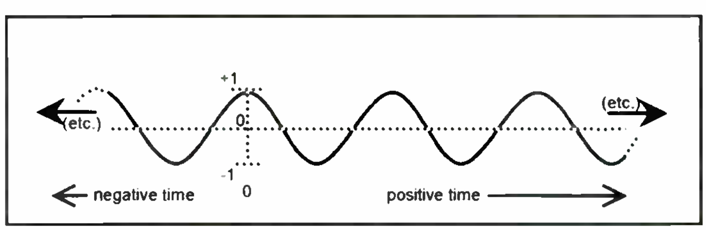
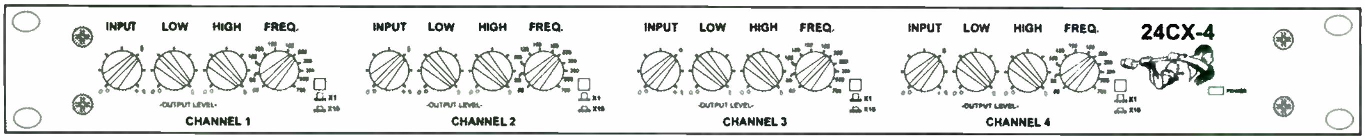


FIGURE 2: A cosine wave oscillates for all time and varies between positive and negative values.



Electronic Crossovers
24CX-2 Stereo 2-Way
24CX-4 Stereo 3-Way

- 24CX-2 is used for BI-AMP and the 24CX-4 is used for TRI-AMP systems or Surround Sound BI-AMP.
- 4th order State Variable Linkwitz-Riley Filter design. All outputs in phase at crossover point.
- 24dB per octave crossover slopes for greater driver protection than with 12db and 18db per octave types.
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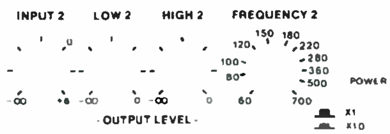
- Zero lobing error (polar pattern tilt) throughout the crossover region.
- Automatic 3-way operation on Models 24CX-2 and 24CX-4 or independent two or four channel 2-way operation.
- Integral security cover.
- Polypropylene capacitors are used throughout the signal path, active filter section, and even the power supply.
- The input/output connections used are high reliability 2mil GOLD phono plug connectors.

24CX-2/4 electronic crossovers represent the latest in state-of-the-art design and manufacturing technology. Innovative electrical and mechanical design concepts and implementation have resulted in a product of superior performance expected of units selling for several times the 24CX-2/4 price.

A 4th order Linkwitz-Riley type of crossover does not guarantee flat summed response. Flat summed response of the low and high frequency outputs of the 4th order Linkwitz-Riley crossover is made possible by using precision 1% metal oxide film resistors, selected capacitors, and the industry's only precision matched and selected 1% four-gang Frequency Range potentiometer. No matter where the control is set, you will get flat summed response, not just the end positions where some products are specified.

High Slew-Rate, Low Noise, Bi FET operational amplifiers guarantee the finest in sound quality and overall electrical performance.



SPECIFICATIONS

Crossover Filter Type
4th Order Variable Linkwitz-Riley design, 24 db/octave slopes.

Frequency Range
x1 60 Hz to 700 Hz
x10 600 Hz to 7.0 kHz

Slew Rate
RL=2 kohms 9 v/usec.
CL=0.01 uFd+/-2.5 v/usec.

Total Harmonic Distortion
RL 2 k ohms

Low Freq Output
.005% THD
20-9 kHz @ +8 dBu (1.95 volts)

High Freq Output
.01% THD
60 - 20 KHz @ +8 dBu (1.95 volts)

Maximum Output Level
RL 2 kohms
+21 dBu (6.2 volts)
@ .05% THD 20 - 20 kHz

Maximum Output Current
25mA peak @ 25oC

Maximum Voltage Gain
+6 dB Hum and Noise
(20 Hz - 20 Khz)
Av = 0 dB fc = 800 Hz

Low Frequency Section
a. Output Attenuater
@ -infinity-104 dBu
b. Output Attenuator
@ 0 dB -96 dBu

High Frequency Section
a. Output Attenuator
@ - infinity 104 dBu
b. Output Attenuator
@ 0 dB 90 dBu

Signal to Noise Ratio
108 db

Input Impedance

Noninverting Unbalanced
20 kohms

Output Impedance
300 ohms

Controls

Input Level:
Continuously variable from +6db gain to 90 db attenuation

Output Level:
Continuously variable from 0 db (unity) gain to 96 db attenuation

Crossover Frequency:
Adjustable from 60 Hz to 900 Hz on the X1 range 600 Hz to 9kHz on the X10 range

Power Source:
U.S. & Canadian Models 120 v AC, 60 Hz

Power Consumption
12 VA

Dimensions (WXHXD)
19 inches x 1.75 inches x 7.5 inches with security cover

Net Weight
both models 8 pounds
0dBu = 0.775 v rms
Also available in 1/4 " phone or XLR Balanced Line

Model 24CX-2 $425.00
Model 24CX-4 $525.00

ORDER INFORMATION: All orders will be shipped promptly, if possible by UPS. COD requires a 25% prepayment, and personal checks must clear before shipment. Adding 10% for shipping facilitates shipping procedure; residents of Hawaii, and those who require Blue Label air service, please add 20%. There is no fee for packaging or handling. We will accept Mastercharge or VISA on mail or phone orders.

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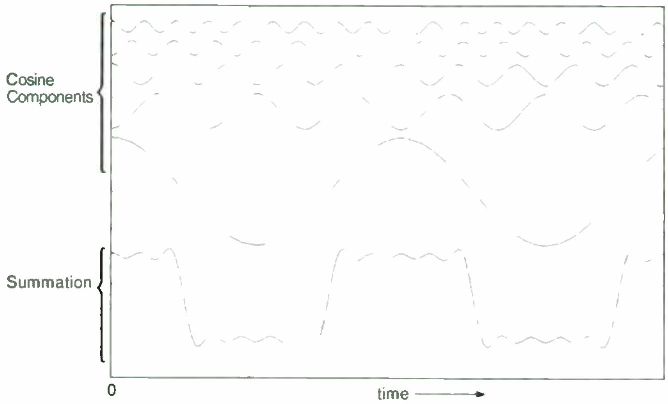


FIGURE 3: Five shifted and sized cosine components summing to form a coarse square wave.

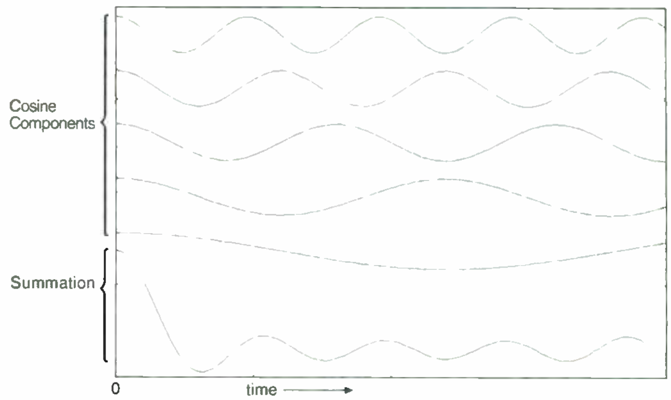


FIGURE 4: Five components of an impulse and their summation. Only at time 0 do they all stack up.

impulse has equal contributions from all frequencies, with all these at zero phase shift (where phase refers to that of a *cosine* waveform). A system that could pass this waveform would have a perfectly flat frequency response in magnitude and phase.

ZERO TO INFINITY

You can envision the construction of an impulse from cosines as follows: since all these cosine waves have zero phase shift, each is equal to 1 at time 0. Therefore, the sum of all the infinite number of cosine waves at time 0 will be the sum of an infinite number of 1s.

At non-zero times, however, some cosine waves (extending up through infinite frequency) have positive values, some negative, and some zero. At times other than 0, the positives will balance out the negatives, with zero as the resulting sum (note that this is not meant to be a proof, but merely a conceptual aid). *Figure 4* shows several equal-sized, zero-phase cosine-wave sections summed to demonstrate how the impulse takes shape.

Because the impulse contains all frequencies phase-aligned at equal levels, an approximation of it makes a very handy test signal. If you feed a bandwidth and size-limited impulse through a system, all its component cosine waves might come out the other side with their sizes changed and their phases shifted. This change is called the frequency response of the system (a frequency-domain representation). The changes to each cosine wave consequently affect the shape of the impulse as it passes through the system, forming what is known as the impulse response (a time-domain representation).

You could apply cosine or sine waves of each frequency, one at a time, to measure the frequency-response characteristic of a system and tally the curve up on a frequency-by-frequency basis. But you can use the *impulse* to do it all at once, assuming you can perform the Fourier transform to analyze the cosine components after they pass through the measured system. A real-world version of the impulse, finite in size and bandwidth, is often used to measure the frequency response of loudspeakers with devices such as the basic IMP.

The reason you go to all this trouble to define any signal as a collection of cosine waves is so you can make unified descriptions of predominantly linear systems such as loudspeakers, which modify signals. If you can describe how a linear device modifies cosine waves of any frequency, you need not measure the separate characteristics of square-wave response, triangle-wave response, or Beethoven's Ninth Symphony response. The full frequency response, including phase and magnitude data, or, equivalently, the impulse response, contains the information needed to mathematically determine how a system will treat any waveform within its dynamic range.

GOING THROUGH A PHASE

The concept of phase tends to confuse a lot of speaker builders and audiophiles, but it's really very simple. The phase shift of a cosine wave is, first of all, a relative measurement defined in relation to an unshifted cosine wave. Scale the size of the wave in question (make it bigger or smaller in height) so that its maximum value is +1. Then see how far to the right or left you must move it to make it match the reference cosine wave.

Phase is always a comparison between two cosine waves of the same frequency. If the reference isn't specified, you can usually assume it to be that at the input of a measured system, or else a spectrum where all cosines are aligned at a reference time. You don't measure phase shift in seconds, but in portions of a cosine-wave cycle. This is the first main difference between phase shift and delay. While delay is measured in units of time, i.e., seconds, the units used for phase shift are radians or degrees (there are 2π radians—or 360°—per cycle).

But because the sine or cosine waves are eternally repeating, a funny thing happens: shifting a wave backward or forward one full cycle of 360° (or any whole number of cycles) gives the same result as not shifting it at all. As a result, you can make any shift possible within a phase change of one cycle. This is so because you are dealing only with cosine waves of a single frequency. As long as you are using a single-frequency waveform for measuring a speaker, you cannot tell whether the shift is x° or (360 + x)°; there is no difference at all between the two.

DELAY SHIFT

Delay, on the other hand, is how far in *seconds* you must shift a waveform to the left on a typical oscilloscope plot to get it to align with a reference unshifted waveform (usually the input signal). If your waveform for analysis is a simple cosine wave, you can still get there by shifting only within whatever time equals 360°.

The actual phase shift corresponding to a given time shift will depend on the frequency of the cosine wave. For instance, if the frequency were 1/360th of a Hertz, each cycle of the wave would last 360s, and a 1s delay would correspond to 1° of phase shift (360°/360). A delay of 6min. would correspond to a phase shift of 360°, the same as no phase shift at all. However, the same 6min. delay for a 1/1440 hertz waveform (each cycle being 1440s), would correspond to 90°.

But most waveforms are not single-frequency waves but are, in Fourier theory, col-

ABOUT THE AUTHOR

Bill Waslo (BSEE, Univ. of Cincinnati) is an RF design engineer with a midwest engineering firm. With his wife, Carol, he also runs Liberty Instruments, developer of IMP, IMP/M and Liberty Audiosuite analyzer systems. His interests include music, loudspeaker design, signal processing, reading, and gardening.

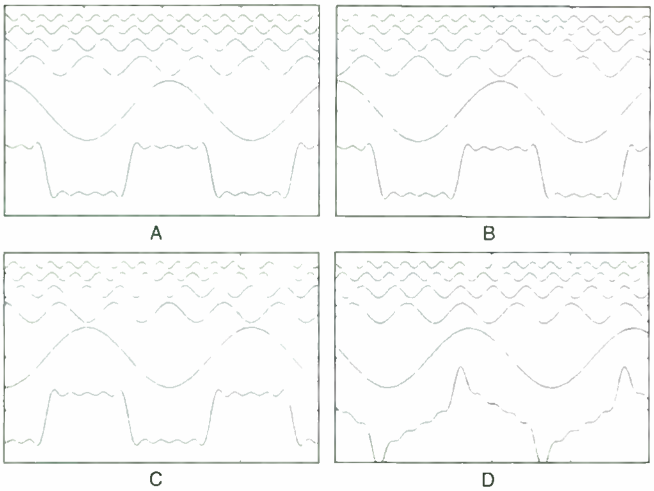


FIGURE 5: The first five components of a square wave and their sum, after various uniform phase shifts (a=0°, b=360°, c=180°, d=65°). Shifts that are not integer multiples of 180° distort the shape.

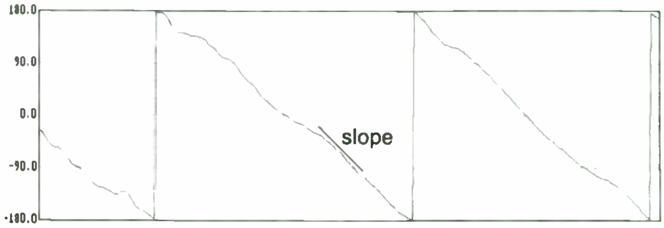


FIGURE 6: A phase response in "linear frequency" format. The group delay at each frequency is proportional to the downward slope of the curve, and is relatively uniform throughout this range.

lections of cosine waves. These complex waveforms may not repeat at all, and if they do, will not repeat at the same rate as each component cosine wave. So delay is not such an ambiguous quantity for such a complex waveform.

Delay, however, can be rather hard to define if the waveform is warped by a magnitude response or is phase-shifted in such a way that its shape doesn't come out resembling the original. This is usually the case for loudspeakers, most of which do not have equal delay at all frequencies (even if the frequency-response magnitude might be essentially flat), consequently altering complex waveform shapes.

This perhaps highlights the reason that the concept of delay is often confusing. In an impulse plot, delay is difficult to determine except in systems which alter waveforms only in horizontal (time) position. Those trying to find the reference plane of a loudspeaker driver by looking only at its impulse response often are unsuccessful because the output doesn't look like the input. Using frequency response and phase data, however, you can easily find delay for one frequency, but only if you already know that it doesn't amount to more than one period of that frequency. Otherwise, the phase at that frequency alone cannot tell you the delay.

PHASE DISTORTION VS. LINEAR PHASE

Whether phase distortion as generated in existing loudspeaker systems is audible is a subject of some controversy. You would probably agree that an imagined pathological system that delayed, for instance, mids and highs by only microseconds, but bass frequencies by several months, would certainly have an audible delay characteristic. But less-severe delay errors may or may not be such a problem, and the dividing line is not clearly known.

If you are designing a speaker and wish it to have ideal waveform replication, what do you look for in the phase response? Do you want a speaker that has the smallest delay possible? One that imparts the same phase shift at all frequencies? One that has flat "group delay"? Or one that shows no curving sections in a plot of phase versus frequency? Just what does an ideal phase characteristic look like?

Delay itself is what audio playback is all about. When you listen to your recording of Belafonte at Carnegie Hall, there is a delay of over thirty years at work on that signal! In speaker measurements, the amount of delay you get depends on the distance from the speaker to the measuring microphone. Remember, sound travels about 1'/ms; you get an additional delay of about 1ms for each foot of speaker-to-mike spacing. In short, the true phase response of a microphone or speaker depends on just how far away you measure it! (It has meaning only if you specify some reference plane.) So there would seem to be little motivation to pursue minimum delay in a speaker design.

How about constant phase shift? You can shift each component cosine wave by a constant phase angle, although the result may not be what you expect. If you shift each component by 360° or any integer multiple thereof, you will, of course, not change any of them and will therefore not change the complex waveform. If you shift by 180°, you will invert the waveform (turn it upside down)! But if you shift by, say, 65°, what do you get?

Figure 5 shows the result of shifting the first five components of a square wave by 0°, 360°, 180° and 65°. Note that the basic shape of the wave is preserved for all except the last. Shifting all the cosine components of a complex waveform by angles that are not integer multiples of 180° usually results in severe waveform distortion! So equal phase shift at all frequencies is generally not desirable from the standpoint of waveform fidelity.

What you want for ideal waveform replication is not a response without delay nor with a constant phase shift versus frequency, but with constant *delay* versus frequency. This is known as a uniform delay or "linear phase" characteristic.

GROUP DELAY

The same delay applied to all frequencies means that the phase shift will be different for different frequencies. In fact, you can express the phase shift of a linear-phase system as being proportional to frequency (remember that each phase angle has many aliases: you can add or subtract any integer multiple of 360° to each angle value without changing it).

A frequency response with uniform magnitude (the "dB" part) and which imparts the same delay to all frequencies will preserve waveform shape. If magnitude or delay is not flat, however, the delay is difficult to determine because the shape of a complex (multifrequency) test signal is changed by the system being measured. And phase shift or delay is rather ambiguous if you consider single cosine waves in isolation.

One way to deal with this impasse is to consider the cosine components at closely spaced frequencies and their phase shifts in relation to each other as they pass through the system. In other words, look at how the phase shift changes for small changes in frequency. This leads to a definition of what is called "group delay," mathematically defined as the negative rate of change of phase versus frequency.

On a plot of phase versus *linear* frequency (*Fig. 6*), this is the "slope" or downward steepness of the phase curve (the sharp upward edges are a result of phase ambiguity where −180° is equated to +180°). A uniform, waveform-preserving phase response will have a constant value of this slope over the entire curve.

But beware! The phase-response curves shown by most measurement systems (including IMP and Liberty Audiosuite) are normally given in log-frequency format. A plot of a uniform delay system, other than one with zero delay, will then show a line for which the downward slope gets steeper as frequency is increased, because more frequencies are scrunched together toward the right side (*Fig. 7*).

In such a plot, a response with a phase-shift line that tilts downward without a bend in it would actually have a nonuniform delay. *Figure 8* shows an overlaid LAUD phase plot for successively increasing uniform delays. The same kind of phase wrapping will occur from the delay that's added by moving a measuring microphone back from a speaker.

Recognize also that constant group delay does not guarantee uniform time delay. A uniform time delay will exhibit constant group delay, but the opposite is not necessarily true. For example, remember the shift of each cosine component by 65° shown in *Fig. 5d*. The phase response of a system bringing about this shift could be represented by a straight line at 65°. The slope of that curve, and therefore the group delay of such a system, is a constant zero. Yet you saw that the resulting waveform distortion definitely indicates nonuniform time delay.

TIME DELAY

A uniform time delay implies that you could remove delay from the phase response to achieve a horizontal line at 0° (which means no delay). If you can't get to a straight horizontal line by removing delay, or if the horizontal line you then achieve is at some angle other than 0°, the system being measured will exhibit waveform distortion. So, it seems that a way to determine whether a delay is constant over a band of frequencies is to work backwards and see whether removing some constant delay will get you to 0° everywhere within that band.

To remove delay at a given frequency, take the time delay being tried (in seconds) and multiply it by 360 times the frequency. Do this for each data frequency and add the result to each raw phase angle (in degrees), remembering that you can also add or subtract 360° as many times as necessary at each point to reduce the answer to within ±180°. Do this for various delay values until a best approximation of a straight line is reached.

That isn't very practical to do by hand. It is much easier to simply apply a known complex waveform to the system and see whether it passes through uncorrupted in the time domain. That is the motivation for square-wave or triangle-wave testing.

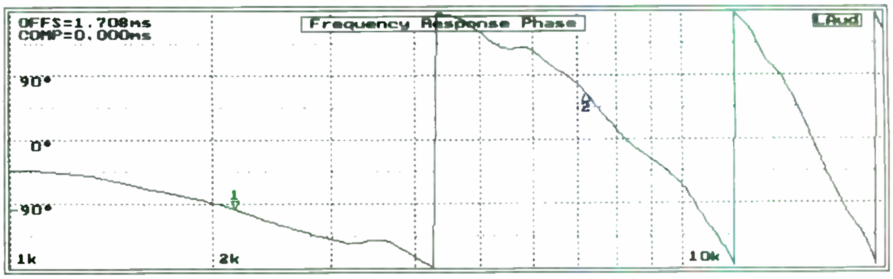
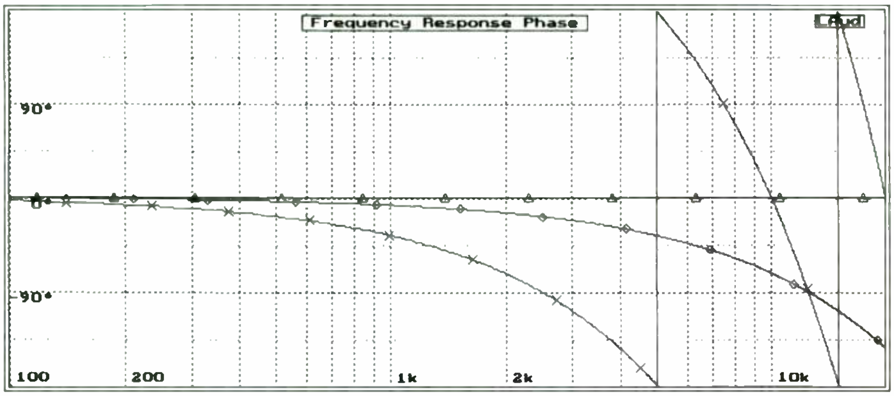


FIGURE 7: The same phase response as in *Fig. 6*, but in log-frequency format. The curve appears to bend more at higher frequencies because of the nonlinear display format, even though the group delay is relatively constant.



Δ ideal phase and zero delay ◊ 20 microseconds delay × 100 microseconds delay

FIGURE 8: An ideal linear-phase characteristic in log-frequency format with different amounts of delay.

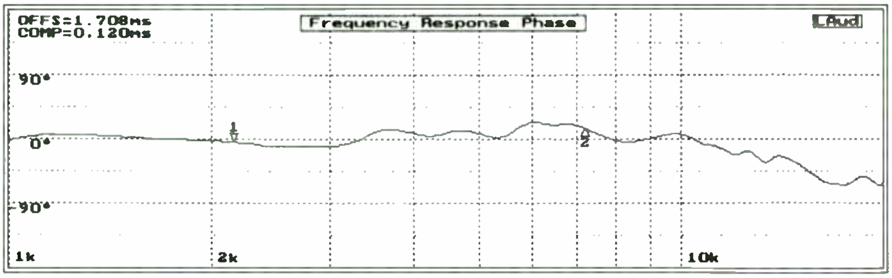


FIGURE 9: The phase response of *Fig. 7*, with time delay ("COMP") removed via computer to resemble a horizontal line. The line is near 0°, so this speaker essentially has a linear-phase characteristic over the range (1–10kHz).

However, even though such simple time-domain testing may tell you that there is waveform distortion, it won't clearly indicate whether any problem uncovered is due to frequency-response magnitude or phase error. In addition, it is difficult to conduct meaningful square-wave tests on loudspeakers, because you cannot remove the effects of the room reflections or echoes that also strongly alter the shapes of the reproduced periodic waveforms.

APPROXIMATION BY COMPUTER

With IMP- or LAUD-generated quasianechoic responses, you can easily remove delay mathematically using the computer until the best approximation to a straight line is achieved on the plot over the frequencies of interest (*Fig. 9*). You can do this by trial and error without much effort. If the line is then essentially near 0° (if at 180°, you can reverse the speaker leads), the delay of the speaker is uniform, and complex waves made up of these frequencies will pass phase aligned.

Of course, the magnitude of the frequency response must also be flat to achieve waveform integrity. In IMP or LAUD, fixed amounts of delay can be subtracted from or added to a phase plot by using the [F9] key.

You can get a close starting value for the amount of delay to remove by measuring the distance from the speaker to the mike and multiplying that value in feet by 0.886 (or the value in meters by 2.91). The result is the number of milliseconds it took for the signal to reach the mike after it left the speaker. For this to be valid (in the case of IMP and LAUD), the measurement should be made with a "Cal" normalization from the signal at the crossover input (in dual-channel mode for LAUD).

You should also set the first time marker to a placement of "1" before doing any transformations, so that the time window includes the entire time of flight. (For further information about normalization, time markers, windows, and measurement devices, see the series of IMP articles in *Speaker Builder* issues 1, 2, 3, 4, and 6 of 1993, the IMP Guide, or the Liberty Audiosuite manual.)

I should again mention that for good sound reproduction you probably do not need such extremely uniform phase-response or waveform-shape integrity. It is not unusual to find that the phase curve of a very good speaker straightens only over short portions of the audio band and often away from 0°, revealing a nonuniform delay. The degree of phase distortion that is tolerable before it affects audible sound quality is another subject altogether.

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D19TD-05-08 ¾″

- High Loss Diaphram
- Magnetic Fluid
- High Power Version
- Good Dispersion

Specifications

Impedance	8Ω
Resonance	1700Hz
Frequency Range	3–20KHz
SPL 1/W/1/M	89DB
Power Handling	80W
Voice Coil	19mm
Magnet	3.7 oz.

$11.69 ea.

P17WJ-00-08 6½″

- Also in 4 ohm
- Magnesium Basket
- Mineral Filled Polycone
- High Damping Surround
- Very Smooth Response

Specifications

Impedance	8Ω
Resonance	37Hz
Frequency Range	5000Hz
SPL 1/W/1/M	88DB
Power Handling	120W
Qts	0.35
Vas	34.7 LTR
Voice Coil	32mm
Magnet	14.6 oz.

$33.93 ea.

D25AG-35-06 1″

- Aluminium Diaphram
- Magnetic Fluid
- Double Chamber
- Flexible Voice Coil Braids

Specifications

Impedance	6Ω
Resonance	850Hz
Frequency Range	1.5–35KHz
SPL 1/W/1/M	89DB
Power Handling	100W
Voice Coil	25mm
Magnet	8.5 oz.

$26.00 ea.

P21WO-20-08 8″

- Magnesium Basket
- Mineral Filled Polycone
- High Temp Voice Coil
- Very Smooth Response

Specifications

Impedance	8Ω
Resonance	28Hz
Frequency Range	25–4000Hz
SPL 1/W/1/M	91DB
Power handling	150W
Qts	0.33
Vas	113 LTR
Voice Coil	40mm
Magnet	24.6 oz.

$47.52 ea.

SHIELDED SPEAKERS

D19TD-03-08 ¾″

- High Loss Diaphram
- Additional Magnet
- Good Dispersion

Specifications

Impedance	8Ω
Resonance	1400Hz
Frequency Range	4–20KHz
SPL 1W/1M	90.5
Power Handling	50W
Voice Coil	19mm
Magnet	7.4 oz.

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M13SG-09-08 5″

- Magnetically Shielded
- Very Smooth Response
- High Damping Surround
- Paper Cone

Specifications

Impedance	8Ω
Resonance	54Hz
Frequency Range	54–5000KHz
SPL 1/W/1/M	88DB
Power Handling	70W
Qts	0.35
Vas	12 LTR
Voice Coil	25mm
Magnet	12.1 oz.

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A MODEST-COST THREE-WAY SPEAKER SYSTEM, PART 2

By G.R. Koonce and R.O. Wright

At the end of Part 1 (*SB* 6/96), I spoke of the design program we used to give us the proportions and detailed dimensions of the enclosures for our three-way system. Now I'll continue with the actual construction.

BOX CONSTRUCTION

Figure 26 shows side and front views of the box. Note that width always refers to a measurement from left to right, height from top to bottom, and depth from front to back. Note also that "box left" refers to the side of the box on your left as you face its front—the same as the position of the "left" speaker of a stereo pair.

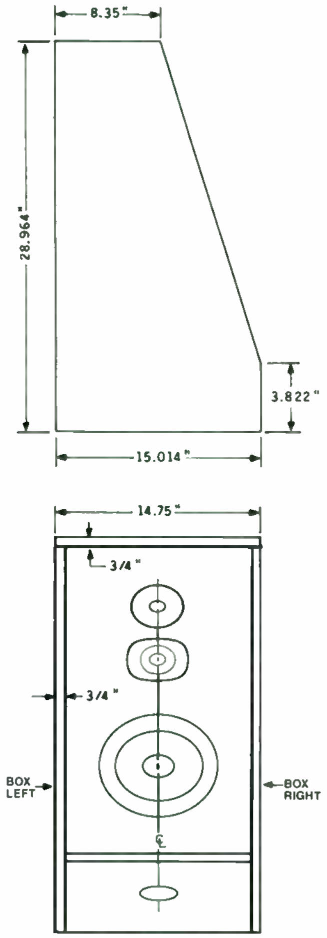


FIGURE 26: Front and side views of enclosure.

The front panel, back, midrange top, midrange bottom, and bottom are all the same width, so if you cut them all with the same saw settings, the box will fit together well even if you don't quite achieve the design width. This is what I call a construction-tolerant design!

I can't carry or easily cut a full 4′ × 8′ sheet of ¾″ particleboard, so I had the lumberyard cut it into 8′-long strips. With the construction of four boxes in mind, I purchased four sheets of the best particleboard and had them cut each one into two strips 13¼″ wide, with a strip about 21+ inches wide left over. It should take less than two sheets of particleboard and some additional pieces of thinner material (for the crossover packaging) to build a pair of these boxes. Guides for designing your own box will be shown in Part 3.

CONSTRUCTION ORDER

I constructed the boxes as follows, working on all four at the same time. Cut the pieces for the front panel, back, midrange top, midrange bottom, and bottom all to the same width. *Figure 27* shows the dimensions of each piece of the basic enclosure. I initially made the midrange top about ¼″ too deep, and also left the midrange bottom long in depth, with only one end cut at an angle for later fitting. I cut the cabinet top and sides later to ensure a better fit.

Photo 3 shows all the final pieces sitting on a side board to help you visualize how the parts fit together. The 2 × 2 had not been installed on the front panel when I took this photo. You should install it after assembling the sides to the front panel and bottom. *Photo 3* clearly shows that both the woofer and midrange chambers have only their sides parallel. It also indicates the very tight tolerances needed on the midrange and tweeter locations to ensure proper fitting of the midrange top.

MAKING THE FRONT PANEL

The next step is to fabricate the front panel. It is easy to get confused here: the end cut at a 90° angle goes at the bottom, and the back (interior) face of the panel is longer. *Figure 28* shows the front-panel layout with all measurements from the bottom, so they are true on both the back and front faces. If your 2 × 2 is not 1.6″ high by 1.5″ deep,

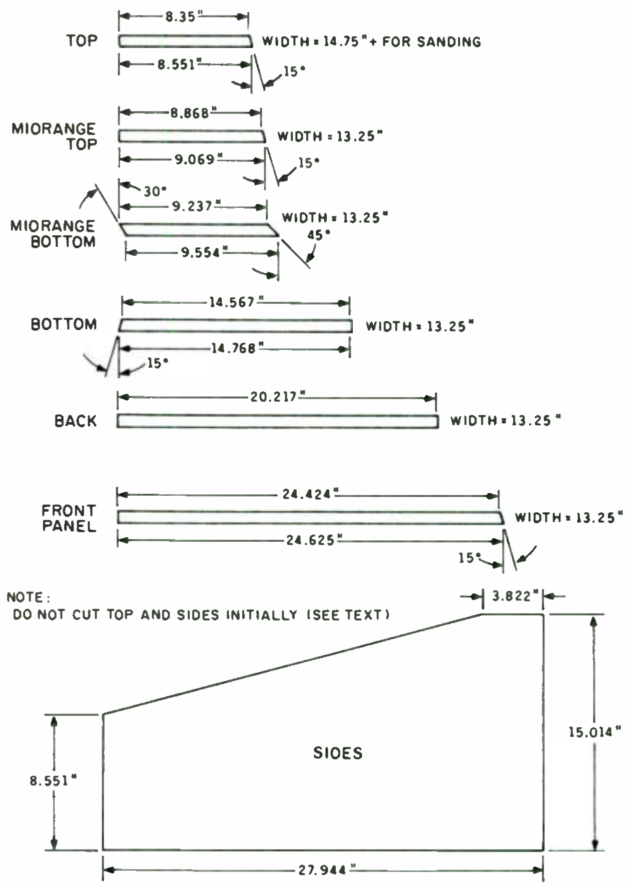


FIGURE 27: Basic drawing of all pieces of ¾″ particleboard in cabinet.

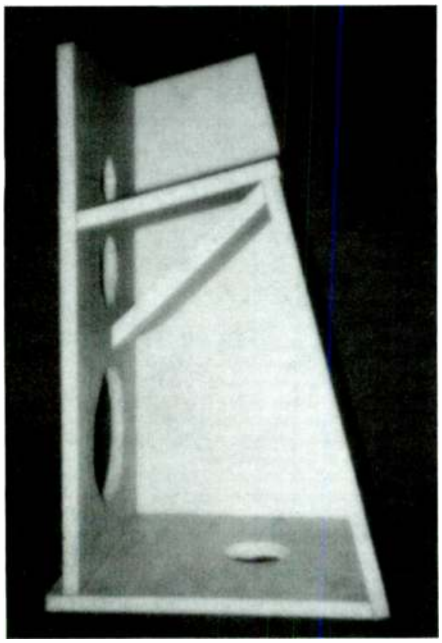


PHOTO 3: Basic layout of panels to construct enclosure.

you will have to make some adjustment.

As noted near the end of Part 1, the tweeter position is raised 1/16″ higher than on the test front-panel design to prevent the back of the rather deep SEAS tweeter from touching the midrange top board. You should check this tolerance later, and, if necessary, grind a slight notch to prevent a possible rattle condition. I did not need to do this on any of the four boxes, but it is tight.

You must mark on the front panel the positions of the three driver holes and of the 2 × 2. Also mark the positions of four nail holes for attaching the midrange top, and four more for affixing the 2 × 2. I assembled my boxes with 1 5/8″ tempered steel panel nails (made by ELCO), which you cannot drive into good particleboard without first drilling holes for them (I used a #51 drill).

If these nails bend, they easily break off, and since they are ridged, they will tear up the particleboard if you try to pull them out. If a nail breaks I recommend you drive the remainder in flush, then drill and install a new nail nearby. Drill the nail holes for the 2 × 2 straight in (at right angles to the surface), but keep in mind that dowels fit into the 2 × 2, so space the nails to avoid the dowel holes (see *Fig. 31*).

You must drill the nail holes for the midrange top at an angle of 15° to the front-panel surface. (You need to be able to drill holes at 15° and 30° angles for this project. See drilling jig in *Photo 8* and discussion later.) My experience has shown that you get better accuracy if you drill the angled holes from the side of the board that mates with the other board. Start the drill vertically to make a small dent (to prevent the drill from walking along the board surface); then set the board to the desired angle and drill the hole.

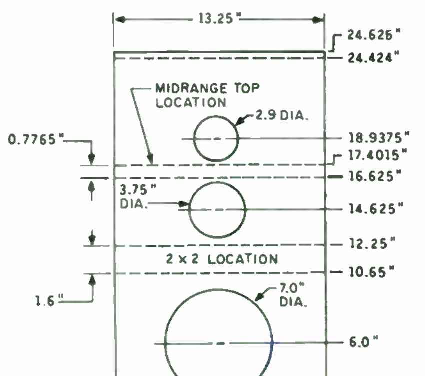


FIGURE 28: Layout of front panel.

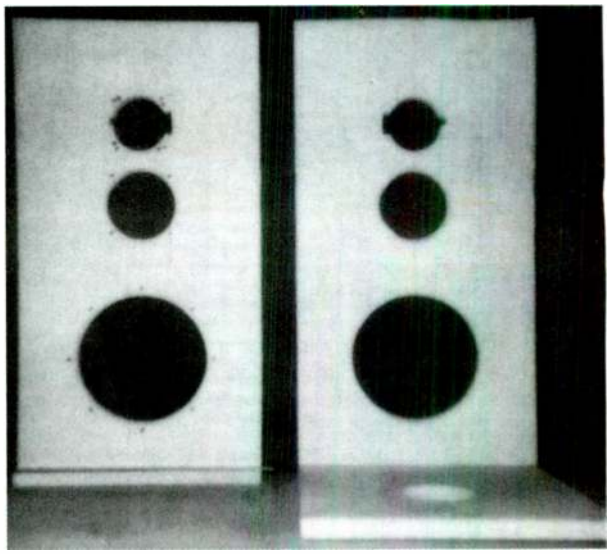


PHOTO 4: Front and rear views of front panel—bottom assembly.

DRIVER HOLES

Next, cut the driver holes using a circle cutter (fly cutter) in the drill press. I do not recess drivers into the front panel, as I feel it weakens this critical mounting interface and is not needed because of the front panel treatment described later. Again, the tweeter and midrange hole locations are quite critical. The tweeter hole must have notches for the terminals, and space gets tight here, because the Audax tweeter requires two notches 180° apart, while the SEAS needs one large notch. There is no room between the tweeter hole and the midrange top board, so you cannot put any of these notches at the bottom of the hole.

You have to be careful here, or you won't be able to properly place the two sets of screw holes needed, as the two tweeters don't have the same screw-hole centers. *Photo 4* shows back and front views of a finished front panel. Mount the SEAS tweeter rotated slightly counterclockwise from having its top two screws on a horizontal line, while you mount the Audax rotated slightly clockwise, with the SEAS notch serving as one of its notches, and the opposing notch cut slightly above the center line. Take plenty of time here. I recommend making a cardboard template that fits both tweeters and then transferring the dimensions to the front panel.

I cut the notches with a Roto Mite cutter and its ¼″ solid carbide bit (#RZ250), both available from Trendlines (American Legion Hwy., Revere, MA 02151, 800-767-9999). I did the final fitting by hand filing to be sure each tweeter type would fit into each front panel. I drilled eight screw holes (#48) for the two tweeter types, mounting them with #6 × ½″ sheet-metal screws, which I find work better in the face of particleboard than normal wood screws. *Photo 4* shows the arrangement of these eight holes.

With the hole sizes shown for the midrange and woofer, it is necessary to file a small chamfer on the front edge of each to get the drivers to fit in flush with the front panel. I mounted both the midrange and woofer with #8 × 5/8″ sheet-metal screws into #43 holes. The wire terminals tend to restrict the rear of the cone, so I mounted both drivers with the terminals facing the top, which is the most restricted area of each chamber. I believe the driver has to "breathe" on the back for the front to work properly, so I always relieve the rear edge of the driver mounting hole. For the sake of maximum stiffness and mounting strength, I do not like to relieve too much material behind the mounting screws and, thus, relieve only the area not blocked by the driver frame. To do this, I set the midrange and woofer into their holes and mark on the back of the front panel the correct areas to relieve.

I perform the reliefs with a router. For the midrange-top relief, I used a 3/8″-radius bit, but did not cut to full depth. You want to set the bit so it will not cut into the area of the front panel where the midrange top attaches. For the other three midrange reliefs, I used 3/8″ radius and full depth. The four woofer reliefs were done with a ½″-radius bit, but not quite to full depth.

Photo 5 is the best picture I could get of

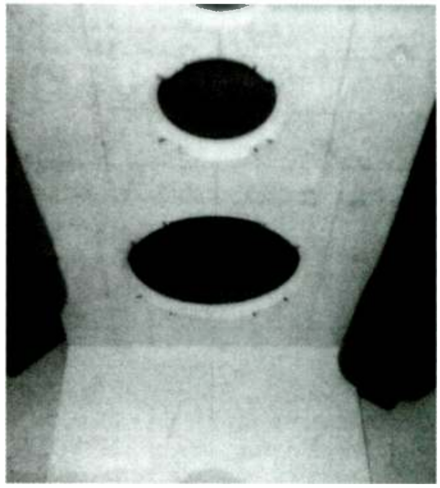


PHOTO 5: Front panel reliefs on midrange and woofer holes.

these reliefs. Building systems with small drivers over many years has shown me that these reliefs are important to get the system to the maximum level of which the drivers are capable, so please do not omit them, even if you must file them by hand.

BOTTOM FABRICATION

The next step is to fabricate the bottom board. Note that the right-angle cut is at the front, and the upper face is the shorter one. Refer to *Photo 3*. *Figure 29* shows the bottom-board layout. Mark the ¾" set-back of the front panel and mark and drill the nail holes to attach this board to the front panel. Then mark the position of the port duct and make the hole. My finished boxes indicate that the diameter of the port duct could be slightly larger (maybe ¼") than mine, but if it's too large, it will end up too close to the back board when you tune the enclosures.

Mating the front panel to the bottom board seems simple to do, but my experience has shown this to be the most critical assembly of the entire box! You must fasten the front panel and the bottom exactly at right angles in two planes. Apply glue to the bottom of the front panel and then clamp the front to the bottom board using a C-clamp hooked into the woofer hole. As shown on

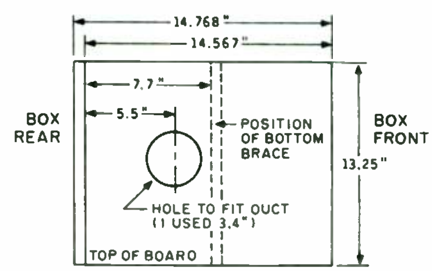


FIGURE 29: Layout of bottom board for enclosure.

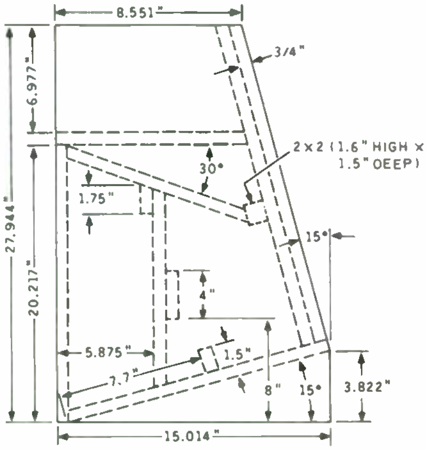
the left in *Photo 4*, verify with a carpenter's square that the front panel is truly at right angles to the side of the bottom. Then nail the two pieces together. By adjusting the position of the C-clamp in the woofer hole, you can "pull" the front panel so you're sure it stands at right angles to the bottom board. Then let this assembly dry.

FABRICATING THE SIDES

You can use the L-shaped front panel and bottom assembly as a template to mark the sides of the box on the wide strip left over from cutting the particleboard sheets. *Figure 30* shows the basic layout of the side boards. Cut out the side boards, making the pairs the same size, even if the boxes are slightly different. Then mark the position of each mating box part (inside and out) on each side board.

Locate and drill the nail holes for the front panel, bottom, midrange top, and back. This includes a hole for a 2″ (coated) sinker nail (#46 drill) into the end of the 2 × 2 that you will mount on the front panel. I did not at this point drill the nail holes for the midrange bottom or the braces that mate with the side, as they are hand fitted and may not come out exactly as shown in *Fig. 30*.

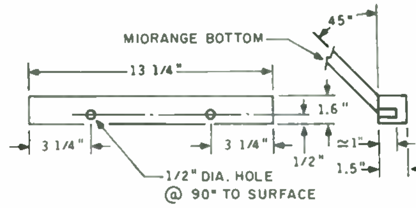
Now glue and fasten one side to the front panel/bottom assembly with just a few nails



NOTE: SHOWS ALL PIECES INTERFACING WITH SIDE BOARDS. ALL ARE 3/4" THICK PARTICLE BOARD EXCEPT 2x2. DO NOT DRILL SIDE BRACE HOLES INITIALLY.

FIGURE 30: Layout of side board for enclosure.

and verify that the back and midrange top will fit properly. Next, glue the second side and fix it in position with long wood clamps. If the box stands properly without rocking, finish nailing the sides to the front panel and bottom board. Drive all the nails in the sides below flush so you can sand the boxes if you wish; what you do will depend on how you plan to apply a finish to the enclosures.



NOTE: MIDRANGE BOTTOM BOARD COVERS ≈1.06" ON 2 x 2 BACK FACE.

FIGURE 31: Details of 2 × 2 brace for front panel.

Figure 31 shows the details of preparing the 2 × 2 braces for installation. Cut them so they fit rather tightly and then install them. Next, set in the midrange top, mark it, cut it to proper depth, and drill the necessary nail holes for attaching it to the back. Pass the midrange wire (#18 Zip cord) through the midrange top to the crossover (CO) area behind the tweeter, choosing a place where no nail from the side board will penetrate the wire. I recommend placing the wire hole 3″ in from the back of the midrange top and 1″ in from its side. I had used a location in ½″ from the side; but, when I got to CO construction, this position caused an interference problem. I knot the wire on the inside and seal it with lots of RTV.

Verifying that the back will fit properly and that the midrange top mates at the correct

Reader Service #32 →

point with the front panel, fix the midrange top in place. Install the wire as noted above, leaving plenty of length each way.

SEAM FILLETS

I advise that you fillet all seams in the construction to ensure air-tight joints and prevent edge buzzing or rattling. I have used many substances for the fillets over the years, but the best I have found is the same glue (Titebond II) I use for assembly. If the joint is not tight, the glue will continue to seep into it until it is. Drying and rechecking takes a bit of time, but you should continue to add a small bead of glue along the joints until it hardens on top, verifying that the joint is sealed.

In placing the glue fillets on the assembled portions of the box, try to avoid areas where braces or other items will later be installed, so you can fit them in more easily. Pay special attention to the midrange chamber, since you will soon close it up, and reaching in through the midrange mounting hole to fillet seams is difficult. When the box is finished, be sure to fillet the seams on the outside of the bottom and front panel.

FINISHING THE MIDRANGE CHAMBER

The next step is to fabricate the midrange bottom board. At the front, this fits against the back of the 2 × 2 brace, flush with its top edge, and against the midrange top just flush with the back board. The front-to-rear ½″ dowels will interfere with the midrange bottom, so you must cut notches in the bottom at the appropriate positions. Also, to attach the midrange bottom to the 2 × 2, drill three nail holes in the 45°-angle cut of the front edge, making these holes perpendicular to the plane of the cut edge. The rear of the midrange bottom is not nailed until you install the back.

Once you install the midrange bottom board, you will no longer have access to the midrange chamber, so the damping material for that chamber must be fabricated before you fasten in the bottom. I was worried about strands of the fiberglass damping material I use getting into the midrange driver via its vented magnet, so I wrapped all the fiberglass mats for this area in old grille cloth. See *Photo 6* to clarify the following description.

Tear 6″ × 17″ strips of (nominal) 3½″ fiberglass into two equal half-thicknesses. Placing two such strips side by side, wrap them in grille cloth, creating a finished 12″ × 17″ mat. Sew the mat closed along the seam and at each end with heavy thread, at one end doubling the grille cloth over and sewing it to make a lip four layers of cloth thick. This is the end you later staple to the front of the midrange top board.

Next, make a cardboard template of the side surface of the midrange chamber, and from this a paper template for measuring the cover of a full thickness 3½″ fiberglass mat cut to fit the side. Then staple the large fiberglass mat to the front of the midrange top. *Photo 6* shows on the left an inverted box with one of the side mats set in. The center box shows the large mat stapled in place. Now poke the midrange wire and the loose end of the large mat through the midrange hole to get them out of the way, and install the midrange bottom.

Use lots of glue on the sides, as it is almost impossible to fillet the inside of these seams through the midrange mounting hole. Now drive the nails into the 2 × 2, clamp the midrange bottom up tight against the midrange top, verify that the back will fit in properly, and then install the nails through the sides. After the glue dries, fillet the midrange-bottom seams, including the one where it meets the midrange top. This last fillet will require you to file clearance on the inside top of the back board so it will fit in properly.

When filleting the seam where the midrange bottom meets the 2 × 2, you need to avoid the holes and notches for the front-to-back dowels. Now, working through the

Reader Service #1 →

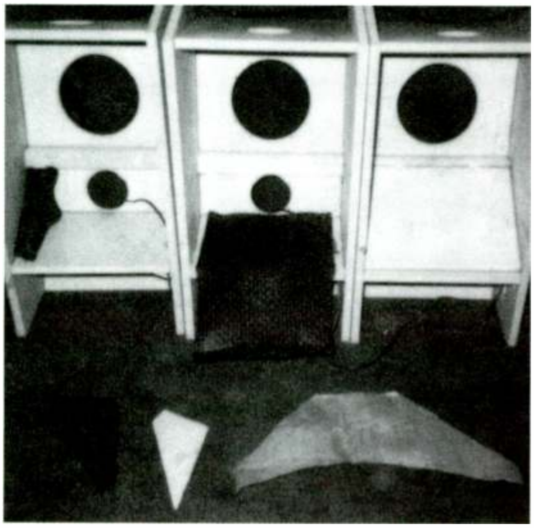


PHOTO 6: Midrange box construction and damping mats.

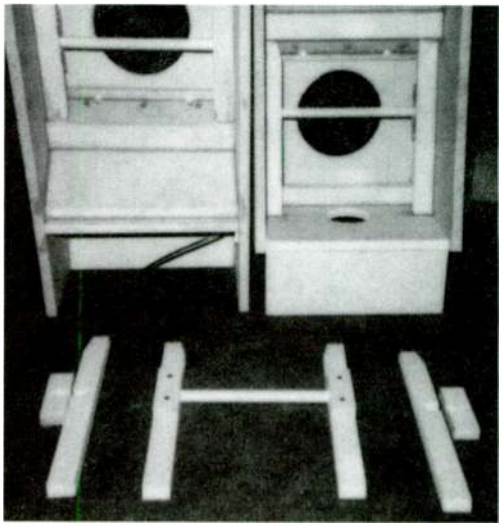


PHOTO 7: Woofer box side-bracing parts.

midrange mounting hole, poke the large fiberglass mat into proper position and put some glue along the lower front edge to keep it in position. Then glue the outer surfaces of the side mats and push them into position. The side mats serve to lock the large mat into position. This finishes the midrange chamber.

SIDE BRACES

The next step is to install the braces that stiffen the top (midrange bottom), bottom, and sides of the woofer chamber (*Fig. 32* and *Photo 7*). To make the two side braces and the back braces, you can use the triangle-shaped sections cut from the side boards to convert them from rectangles to their final shape. The braces will not be full width for their entire length, but this is no problem. Install the bottom brace first, ½″ in front of the port hole and perpendicular to the bottom board. Drill holes and drive nails into this brace through the bottom and sides. The remaining braces do not mate with this one.

Fitting the side braces is painful hand work. They do not have to be in the exact position shown, but you don't have a lot of clearance between the side-to-side dowel and the woofer magnet. The braces go in vertically, parallel to the back. Once the side braces are cut to fit and you've made the top brace with the two #36 holes drilled, install all three pieces, clamping the tops of the side braces to the top brace. Then drill the two #36 holes into the midrange bottom board, mark the dowel location 10″ up from the bottom of the box, and mark the brace locations inside the box.

Now remove the pieces, drill the dowel holes and the dowel clamping-block-attachment screw holes, and put the through-bore and countersink on the top brace. The flat Irwin bits will drill the holes for these dowels in particleboard with no problem, straight in or at an angle. These holes, about 1″ deep, are at the interface of the brace and the clamping block. Drill the necessary nail holes in the box sides and then install all three parts permanently, along with the ¾″ dowel (about 12″ long) and the dowel clamping blocks.

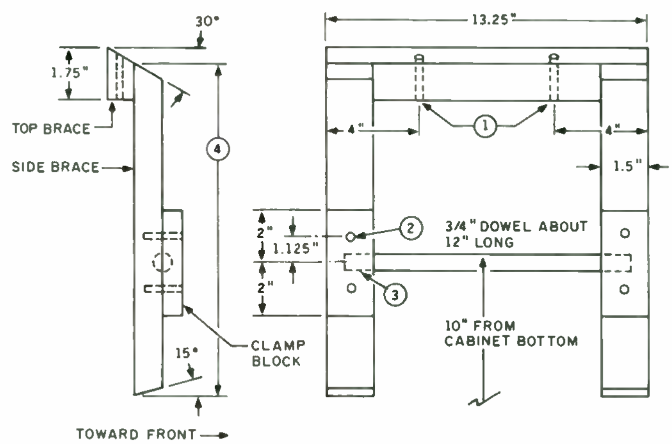
BACK BRACES

The back braces are a bear. *Figure 33* shows the dimensions, with the layout shown in *Photo 8*. Put the dowels in the 2 × 2 and mark the dowel-hole locations on the fitted back brace—about 10 7/8″ up from the inside bottom edge of the boxes. Remember that these dowel holes go in at a 15° angle. Once you've finished the back braces and dowel clamping blocks, cut the ½″ dowels to the proper length (about 8 11/16″) and assemble all parts in the boxes.

Mark the positions where the braces fit best on the midrange bottom and the box bottom—about 3¼″ from the inside edge of the side boards. Drill nail holes in the bottom board and toe-nail holes in the 30°-cut faces of the back braces at the top. Then install both back braces, making sure they will sit tightly against the back board when it is installed. Be sure to drive the toe-nails at the top in flush so the back will fit. Also, file grooves along the sides of the ½″ dowels where they go into the 2 × 2, so glue will not prevent their insertion.

After the glue has dried, fit the back board in temporarily. I did not think I could support the back braces sufficiently to drive nails into them, so I drilled three holes for #6 × 1½″ flat-head particleboard screws in each side of the back. You should also mark the back for nails into the bottom board (angled up at 15°) and into the midrange bottom board (down 30°), being careful to avoid existing nail locations.

Photo 8 shows an upright and an inverted box with the back braces installed. Also shown at the left front of the picture is the 15° drilling jig for the drill press. The remov-



NOTES: (1) DRILL INITIALLY #36. IN FINAL PREPARATION DRILL TO #12 AND COUNTERSINK FOR #8 x 2" FLAT HEAD PARTICLE BOARD SCREW.

(2) INITIAL DRILL #36. DRILL CLAMP BLOCK #13 AND COUNTERSINK FOR #8 x 1 1/4" FLAT HEAD PARTICLE BOARD SCREWS.

(3) 3/4" HOLES FOR DOWEL DRILLED APPROXIMATELY 1" DEEP.

(4) SIDE BRACES CUT TO FIT; OVERALL LENGTH IS ABOUT 14" SO START LONGER. IF BRACE NOT FULL WIDTH FOR ENTIRE LENGTH PUT NARROW END AT TOP.

FIGURE 32: Details of all pieces involved in side bracing woofer box.

PHOTO 8: Woofer box back braces and damping material.

able dowel across the top of the jig allows drilling at 30° when the board is properly positioned. Also shown is a back board with the wire installed on the cabinet's left side, down 3″ from the top of the board and in 1″ from the side.

THE WOOFER BOX

We were unable at the time to obtain a second pair of woofers, so from this point construction continues on only two boxes. Line the woofer boxes with fiberglass as follows. Cover the bottom board with nominal 3½″ fiberglass in three areas: in front of the bottom brace, behind the bottom brace, and outside the back braces, but here stopping about 2″ from the back of the box. This leaves the port area clear, and the gap at the rear allows room to place fiberglass on the back.

Cover each side with nominal 3½″ fiberglass both in front of and behind the side braces, but again leave about 2″ from the cabinet rear uncovered.

At the top of the woofer box, cover the bottom side of the 2 × 2 with ½″ fiberglass, wrapping it around the 2 × 2 up over to the dowel area. (This fiberglass is really ½″ thick and is sold as wrap-around insulation for pipes.) Then use the nominal 3½″ fiberglass from the 2 × 2 to the top brace and from that brace back to 2″ from the cabinet rear. Cover the fiberglass from the top brace to the front panel with grille cloth to keep fibers from getting into the woofer.

The right side of *Photo 8* shows the box at this point of construction. Next, cut three strips of fiberglass to cover the back board (avoiding the back braces), but do not install them yet. Now drill the needed holes in the back board, including countersinks for the screw holes, glue the board and nail and screw it in. (Don't forget the woofer wire before you install the back.)

With the box on its back, reach in through the woofer hole and fillet the inside back seams (the back is later filleted on the outside also). With the interior fillets done, install the last three strips of fiberglass through the woofer hole, poking them under the existing fiberglass on the sides, top, and bottom to hold them in place while the glue sets. (I initially used rubber cement to glue the fiberglass, but we discovered it came loose when the boxes were shipped. You should use a stronger glue or, alternatively, the approach described by R.O. Wright in Part 3.)

CROSSOVER PARTS

At this point, the box is complete except for the CO parts and the top board. *Figure 34* shows the dimensions of the parts for the Zobel board, and *Fig. 35* the parts to fabricate the CO board. The two 15/16″ holes in the rear of the Zobel board (*Fig. 34*) serve as finger holes to permit handling and to allow

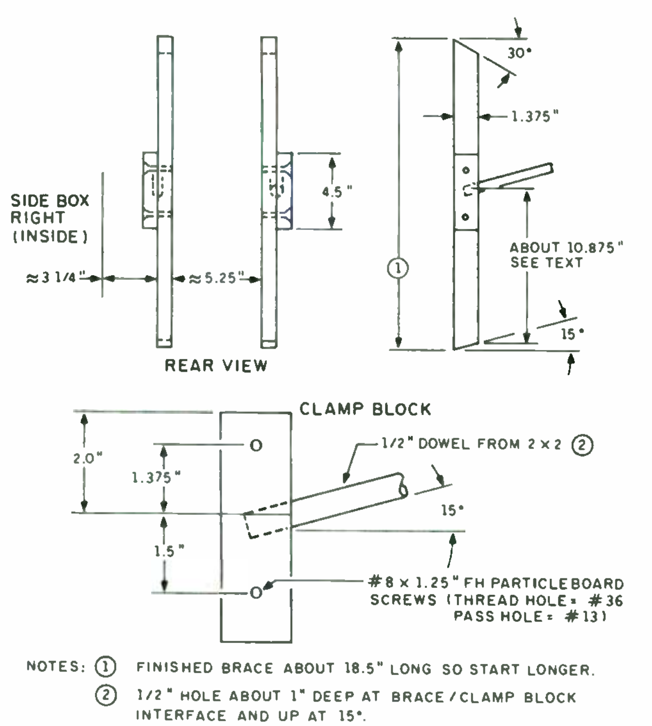


FIGURE 33: Details of all pieces involved in back bracing woofer box.

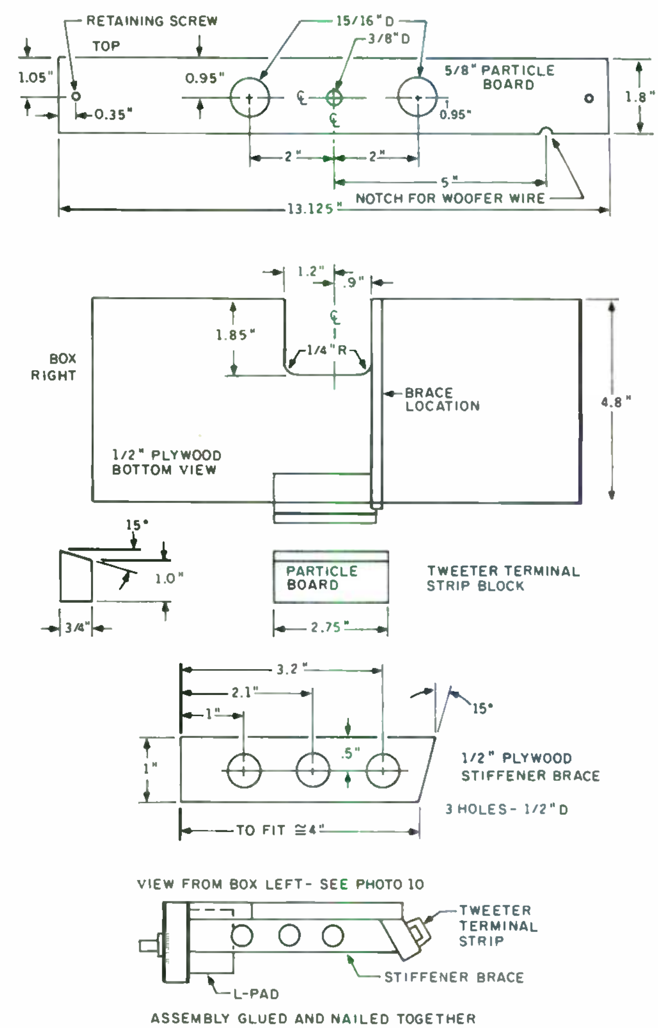


FIGURE 34: Details of all pieces used to construct the Zobel board.

air flow, as this board holds several resistors. Do not omit these "finger" holes or the holes shown in the Zobel-board stiffener.

Photo 9 shows the pieces of the boards. At the right front are the three pieces that build the Zobel board (I added a fourth stiffener piece later), and at center front are the two pieces that form the T-shaped CO board, of which two are needed per box. On the left are finished Zobel and CO boards, with the only components mounted being the L-pad and tweeter terminal strip on the Zobel board.

The E-shaped pieces (*Fig. 36*) are screwed to the enclosure sides to hold the boards as shown in the box at the right rear of the photo. Note the foam tape covering the groove faces. The box at the left rear shows how it appears with the boards installed.

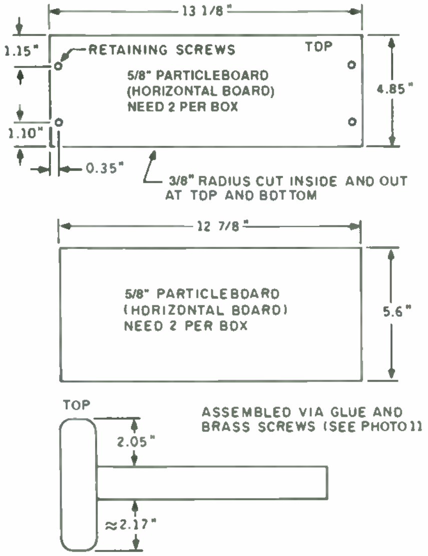


FIGURE 35: Details of all pieces used to construct the crossover boards.

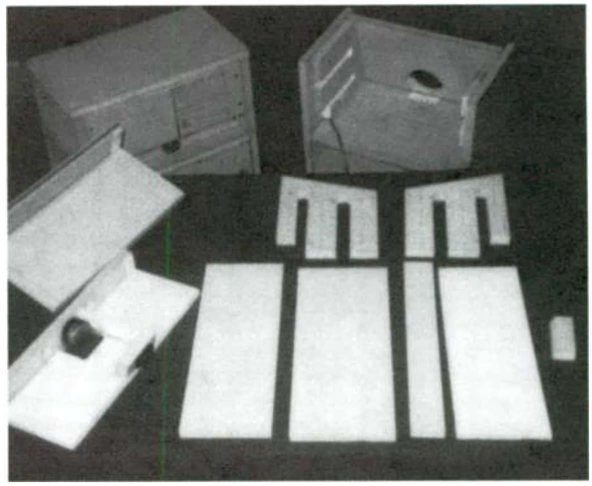


PHOTO 9: Pieces used to package and mount crossovers.

TIGHT QUARTERS

The approach does provide easy access to the tweeter L-pad and a quick and convenient way to change the tweeter and CO. If the distance from the upper surface of the midrange top to the upper edges of the sides is not at least 6.9″, you may never get the COs packed in this area and should use a different approach. Even with 6.9″ available, the space was so tight that I was forced to make the horizontal portion of the Zobel board out of ½″-thick plywood rather than 5/8″ particleboard. The plywood was then so lively that I added a stiffener from the vertical board to the small block for the tweeter terminal strip to damp the board so it would not ring.

After fabricating the parts needed for packaging the COs, cover the slide grooves in the E-shaped holders with 1/16″ single-sided foam tape, including the front edge of the short (bottom) E leg. Place additional foam tape on the inside faces of the vertical boards, on both the Zobel and CO, to make

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sure there will be no contact between particleboard pieces to cause a buzz problem; this is clearly visible in *Photo 9*. Then clamp the E-shaped pieces into position on the box sides and verify that the boards fit without interference.

Finally, screw the E-shaped mounts into the sides. This violates one of my basic rules, as they were not glued or filleted, so a potential for buzzing existed. I wanted to be able to move them if need be, and if a buzz resulted, I planned to reinstall them with a thin sheet of damping material clamped between them and the cabinet sides. No problem developed here, however.

FINISHING THE BOX

You can now cut out the top board, making it a bit wider than the measured dimension so you can sand it smooth. Drill holes for nailing the top to the sides and front panel (at 15° angle), then glue and nail it on and fillet it on the inside. Drive the nails in to below flush. The construction is now complete. Now, you should do whatever is needed to finish the box. I sanded the outside faces and painted the bottom diffuser area with flat black paint.

Builders have finished boxes of this type by painting them, veneering them, laminating them with Formica®, or wrapping them with stick-on vinyl. As for grille cloth, this design intends for you to staple it right on the front of the box and cover the staples with thin trim strips, but this would prevent tweeter changes. If you want to be able to change the tweeter, you can fasten the grille cloth to a very shallow removable frame.

Also, if you object to the exposed particleboard edges at the bottom, you can install a bottom board. I have experienced no sonic change with this, but it makes future retuning of the woofer very difficult unless the board is removable or contains a hole aligned with the port.

DRIVER INSTALLATION

Solder the woofer and midrange to the preinstalled wires and fix them in place using Moretite putty-like material to obtain an airtight seal. These drivers are then removable in case you want to play with the damping material in their chambers. To the tweeter, solder wires of sufficient length to allow it to sit on the top of the box and still be connected to the terminal strip on the front edge of the Zobel board. These pigtails should terminate in space lugs, as you will not want to have to remove the terminal-strip screws to change tweeters. The tweeters do not need an air-tight seal to the front panel, so you can install them dry.

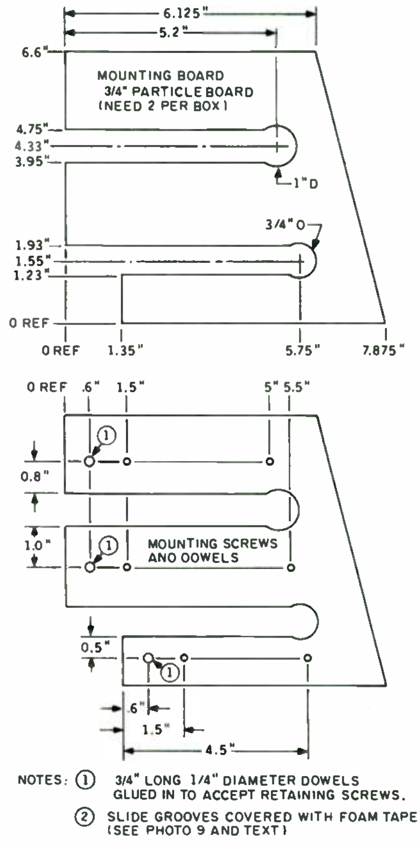

FIGURE 36: Details of the E-shaped pieces.

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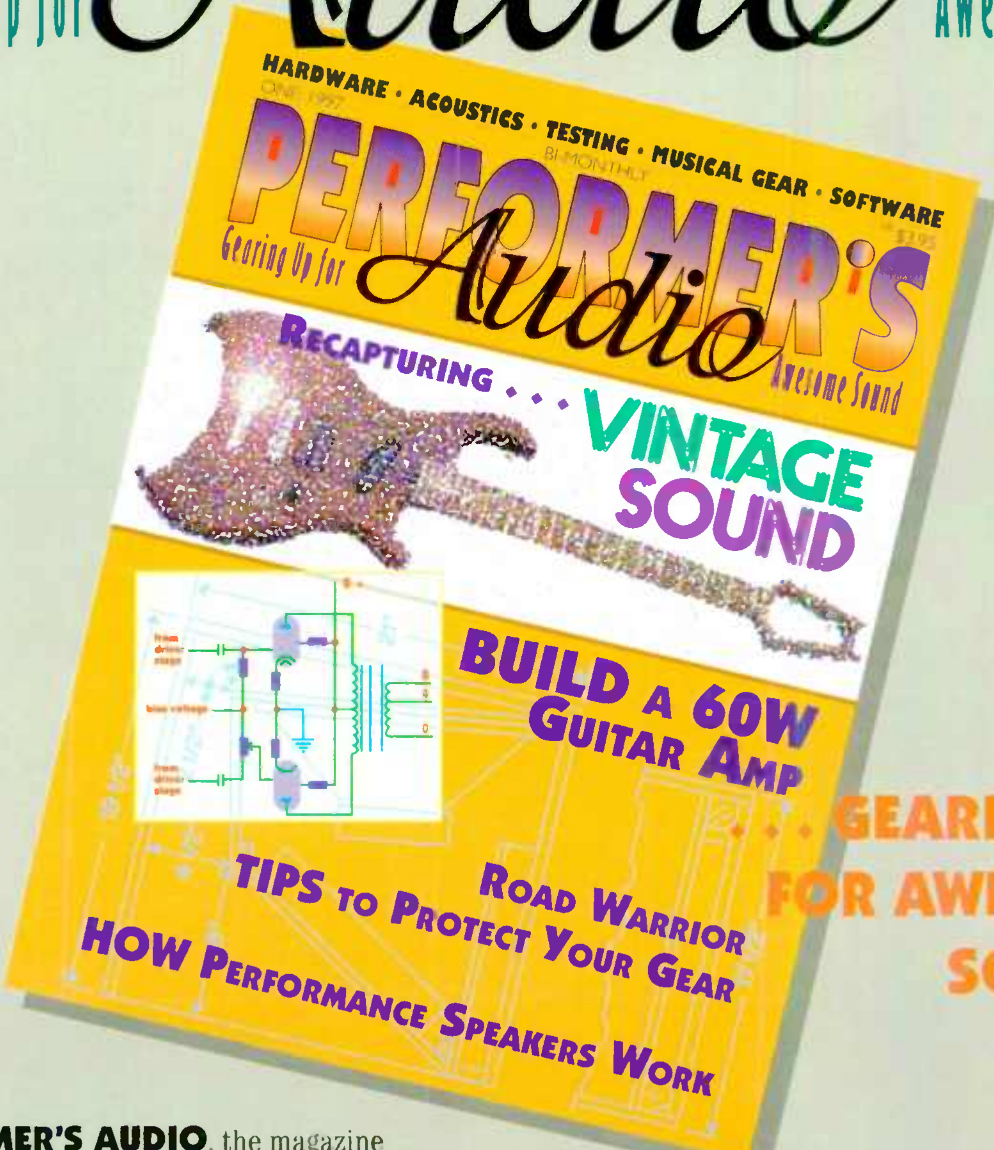

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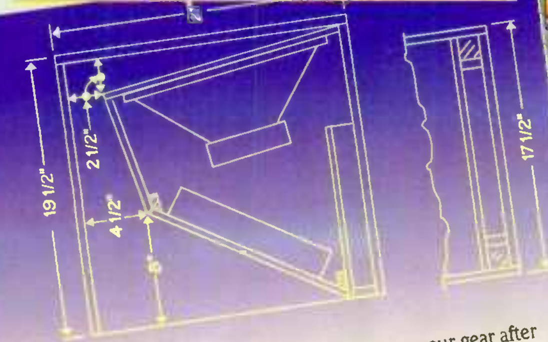


It's 2 a.m. You've just finished packing up your gear after a gig and are heading out to your pickup truck in the parking lot. It's raining. The drive home is 30 miles, and you pray that your bass bins won't be totally waterlogged before the night is over.

The drums have cases for protection, and your electronics are safely sealed in roadcases, but what about your speakers? At best, all that separates your costly drivers from the elements are flimsy vinyl covers.

If you'd like to change this situation, I can show you how to build good roadcase protection for your speakers.

(From **The Road-Warrior Subwoofer**, by Bill Fitzmaurice)

I discovered a cheap & easy way for guitar players to sample single-ended sound with almost any output and power transformer. In my design, the output pentode runs single-ended "ultralinear" mode. The center tap of the output transformer's primary is connected to the screen for a louder guitar amp.

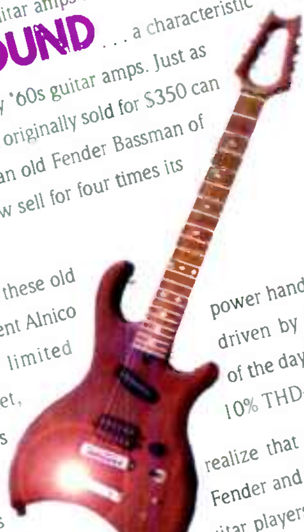
Single-ended musical instrument amps have a "musicality" that is great for guitar but horrible when overblown in playback (hi-fi) amps. The sound of this amp is not hi-fi. It adds a dark, pleasant thickening from significant amounts of harmonic distortion, no feedback, and iron distortions (with apologies to Kondo San, iron is bad).

In this article, I help you choose cathode and plate resistors for both guitar and hi-fi. You'll have the ability to kludge any amp.

(From **Guitar Amp Refinements**, by Daniel Patrick Coyle)

SAMPLE ARTICLES FROM THE PREMIERE ISSUE

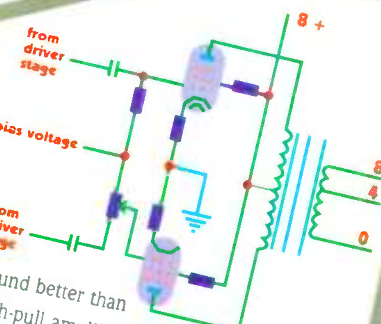
hottest trend in today's guitar amps is the **VINTAGE SOUND** . . . a characteristic of the late '50s and early '60s guitar amps. Just as 1959 Les Paul Gibson that originally sold for $350 can now fetch up to $25,000, an old Fender Bassman of the same period might now sell for four times its original price.



What was the magic of these old amps? Imagine inefficient Alnico magnet drivers of limited power handling tossed in an open-back cabinet, driven by an antiquated (even by standards of the day) tube amp that ran with about 10% THD—on a good day.

Few musicians realize that the two giants of guitar-amp design, Leo Fender and Jim Marshall, were neither audio engineers nor guitar players! Technically speaking, their amps were crude affairs incapable of producing clean, full-range response. But while their inherent distortion and speaker coloration made their amps worthless for quality sound reproduction, those same sonic shortcomings applied to an electric guitar's signal were responsible for defining the sound of a generation.

(From **Build Your Own Guitar Mini-Cab**, by Bill Fitzmaurice)



For a-long-time-now ...we have known that single-ended guitar amplifiers sound better than push-pull ones. This is because a push-pull amplifier produces only odd harmonics when clipping. Most guitar amplifiers are built as push-pulls because single-ended transformers are heavy and expensive.

You can very easily modify a push-pull amplifier to give a single-ended characteristic. By removing the signal from one of the output tubes, you can convert your amplifier to single-ended. One output tube works as an output tube and the other works as a current generator to prevent the output transformer from saturating. With this simple modification, you can continually change your amplifier from push-pull to single-ended operation.

(From **Modify Your Push-Pull Guitar Amp**, by Rickard Berglund)

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Julian J. Bunn

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The AIRR software includes operating instructions combined with on-line help and a sample file containing microphone response data. AIRR requires a PC equipped with a SoundBlaster 16 soundcard (or compatible); VGA screen with at least 640 x 480 resolution; microphone (usually supplied with the sound card, although a calibrated microphone such as Old Colony's Mitey Mike is often desired); and loudspeaker to test. Math coprocessor not required. Shipping weight: 1 lb.

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Visual Ears allows the audio enthusiast to enhance system performance by minimizing standing wave and boundary reflection problems with proper speaker and listener placement. In most cases, room problems can be reduced to satisfactory levels, at a prime listening position, without resorting to obtrusive acoustical control devices. The program features many enhancements:

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* Uses high speed 32-bit algorithms written in C.
* Models system low frequency roll-off.
* Saves multiple set-ups in a single file.
* A multiple file capability is useful for dealers who are setting up customers' rooms.
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Requires Windows 3.1 or 95, VGA monitor and mouse.
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Sitting Duck Software

This interesting program predicts standing wave modes in small rooms and is designed for positioning speakers—and the listener–in such a way as to minimize standing wave effects and other room-generated influences. It allows for a full range of speaker and listener movement in 3D space and continuously updates a standing wave Pressure versus Frequency display.

IBM 256K RAM; DOS 2.11+; CGA, EGA, VGA, or Hercules graphics required.
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The Listening Room For MAC
Sitting Duck Software

Like the popular PC version, this program for the Macintosh allows you to interactively position listener and speakers in your listening room, although this is now accomplished using the Macintosh "dragging" graphics technique. The program actively displays the magnitudes of standing waves, as well as the effects of (now up to 124) early reflections on the direct response. Several features have been added, notably local optimization of listener and/or speaker positions to maximize, minimize, or smooth the standing wave patterns. This MAC version also enables you to model the woofer low frequency limit and slope, and produces high-resolution output including multiple graph overlays. Requires 512k RAM. Shipping weight: 1 lb.
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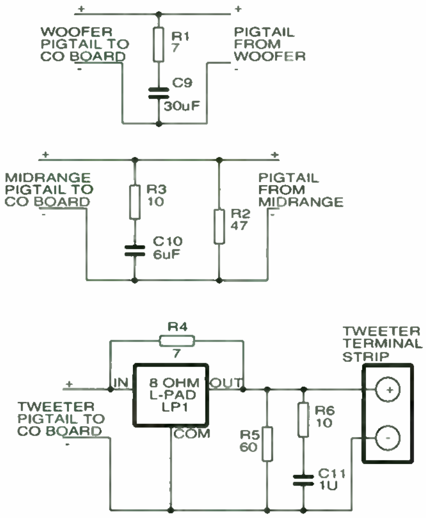


FIGURE 37: Schematic for components packaged on Zobel board.

WOOFER TUNING

Next, tune the woofer. Start with a duct tube 8″ long and proceed to cut it off until you obtain the desired impedance-curve minimum (f_m), assuming this should match the computed box-tuned frequency (f_b). Remember to stand the box up on the floor when checking the tuning; standing it near the edge of a workbench will not yield the correct results.

For my woofer A, the desired f_m was 38.6Hz, and the final duct was 5¼″ long. For woofer B, the values were f_m = 37.6Hz and length = 6.0″. If you do not plan to measure your drivers to establish an f_m value, I would go with a 3″-diameter duct about 5.5″ long. (The second pair of boxes with the newer woofers required 3″-diameter ducts of 6½″ and 6¾″ length and I now recommend a length in this range.) Note that setting f_m typically does not result in quite the desired (same) f_b.

The system performance just does not change that quickly with small tuning variations, which is good, since environmental variations will move the tuning around anyhow. If you insist on spikes on the bottom of the box, they should be in place during the tuning process. I have no idea whether they would affect the sound, as I have never tried lifting the diffuser off the floor.

TABLE 4

ZOBEL BOARD PARTS

(All capacitors are plastic-film units of 100V or greater.)

UNIT	VALUE	HOW PART WAS IMPLEMENTED
C9	30µF	Parallel 10 + 10 + 10
C10	6µF	Parallel 5.6 + .47 (selected)
C11	1µF	1µF selected
R1	7Ω	5Ω @ 10W series 2Ω @ 11W
R2	47Ω	47Ω @ 5W
R3	10Ω	10Ω @ 7W
R4	7Ω	3Ω @ 5W series 4Ω @ 5W
R5	60Ω	47Ω @ 5W series 15Ω @ 10W (selected)
R6	10Ω	10Ω @ 7W
LP1	8Ω	L-pad marked 15W, 1.64″ diameter, ¾″ threaded mounting collar.
TS1	2 terminal	Terminal barrier strip—#5 screws, overall 1.4″ long × 0.85″ wide.

DAMPING THE FRONT PANEL

The ¾″-deep grille frame and covering grille cloth can have adverse effects on the clarity of the system's high-frequency response; detailed discussion and test results will be shown in Part 3. Covering the front panel with damping material greatly reduces these effects and removes the need to attempt to set the drivers flush into the front panel. I, thus, cover the front panel with ½″ fiberglass, the 6″-wide kind made for wrapping pipes. *Photo 12* shows a front view of the enclosure with this covering.

ZOBEL-BOARD PACKING

At this point in construction, the box could be played, so I breadboarded the COs and tried them. I was happy with the sound; no changes were needed in the CO values developed via the baffle and breadboard tests.

Wanting to proceed slowly, I packed only the Zobel-board components. *Figure 37* shows the schematic for the Zobel board, and *Table 4* lists the component values. *Photo 10* shows the layout I used in trying to maintain minimum inductance in the wiring. The pigtails should be long enough to allow pulling the CO board all the way out when they are attached. The pigtails end in terminal lugs for attachment to the barrier terminal strips on the CO boards. I used spade lugs so I could disconnect the wires without removing the terminal-board screws.

I played the systems with the installed Zobel boards and the COs still in breadboard fashion. I could detect no difference in the sound quality. Note that a notch must be filed in the bottom edge of the Zobel board's vertical face to allow clearance for the woofer wire. I soldered the woofer and midrange wires directly to the Zobel board, as there is no reason to make it removable.

CROSSOVER BOARDS

Next, I packaged the two CO boards for each system. *Figures 38* and *39* are, respectively,

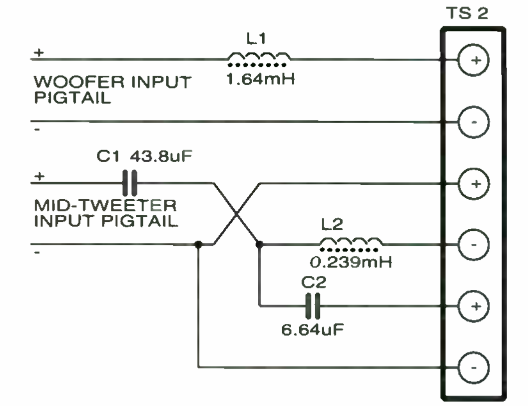


FIGURE 38: Schematic for components packaged on first-order crossover board.

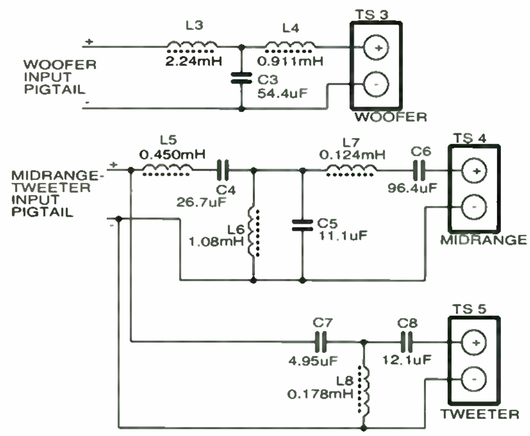


FIGURE 39: Schematic for components packaged on third-order crossover board.

the first- and third-order CO-board schematics, and *Table 5* lists the component values for both. I used large film capacitors for these COs for two reasons: I wanted to be sure there was enough room for others to build the CO, and I thought from past experience that the systems sound better with the physically larger capacitors.

Photo 11 shows the top and bottom views of the two CO boards. The black stripe on the mounting board indicates the top of the structure. The first-order CO is easy to package, and I mounted all components on the top of the horizontal board to keep them as far as possible from the tweeter and the large metal shell of the L-pad. I used a single six-terminal barrier strip for this CO board to provide terminations for the three pigtails from the Zobel board. I used two input pigtails, one feeding the woofer and the other the midrange and tweeter to allow biwiring if desired. The length of these pigtails depends on where you plan to locate the amplifier input terminals.

Again, I recommend minimum-inductance wiring practice; see *SB* 5/90, p. 26, for information on this. Remember that with the first-order CO the midrange is wired so that when the pigtail is connected, the midrange polarity is inverted.

The third-order CO board is a bear. I spaced the coils as best I could to keep them away from the coils on the opposite side, from the tweeter, and from the L-pad on the Zobel board right below the CO board. With the large film capacitors, this board gets full. The woofer LP and tweeter HP are on the bottom of the board, and the midrange BP occupies the top. Again, use minimum-inductance wiring practices and the two input pigtails. The pigtails from the Zobel board terminate on three individual two-terminal barrier strips, the woofer and tweeter on the bottom and the midrange on the top.

CROSSOVER PROBLEMS

The first-order CO board was installed, and there was no change in the sound as far as I could determine. That was not the case when I tried the third-order CO board. I noticed a slight increase in sibilance, generally caused by an excess of energy in the 4kHz range, which is the upper CO frequency. Certainly the speakers were not perfect before this, but I believe I heard a change, and I now preferred the sound with the tweeters set about 1dB lower in level.

I examined and measured the third-order CO boards, but all components were correct and were the same ones used in the breadboarded CO listening. ROW reports that coil crosstalk will cause the effect observed, and this may indeed be the problem, but I have measured side-by-side ferrite bobbin core coils and found them rather immune to this. I tried sliding the CO board back to see if the L-pad was interacting with a coil (these coils are more sensitive to interaction with items off the end than off the side), but that made no difference. I pursued the mystery no further.

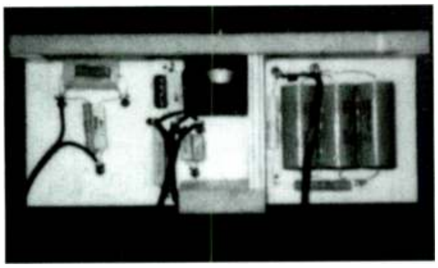


PHOTO 10: Bottom view of finished Zobel board.

For the best sound, I recommend that you do not pack the third-order CO as I did. There was also cabinet-top vibration, which is not good. This is a result of the CO packaging not providing room for a stiffener on this board. If you were to package the first-order CO on the bottom of the board (keeping inductors away from the tweeter and L-pad) instead of the top, you could put a brace on the cabinet top board. But you would then have to package the third-order CO outside of the box.

Perhaps a design with a pedestal at the bottom to house the CO would be a better approach. I plan to investigate this problem on the second pair of boxes and will report a solution if I find a good one. I also think more experimentation with the third-order CO component values could further smooth performance in the region of the upper CO frequency. (Follow-up work in this area will be reported in Part 3.)

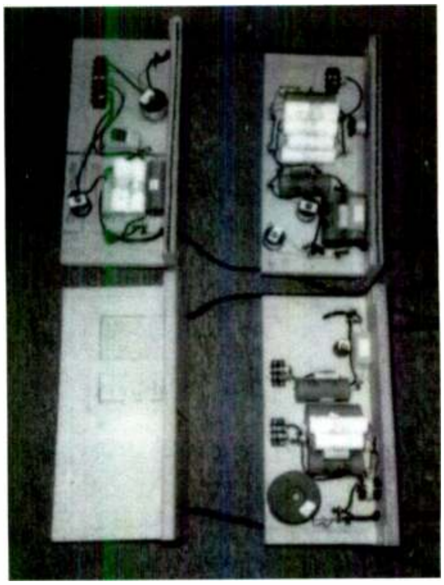


PHOTO 11: Top and bottom views of finished crossover boards.

Photo 12 shows a front and back view of the finished boxes. Amplifier input is via a barrier terminal strip mounted on the back of the box. I provided a jumper to tie together the two CO pigtails for single-wire drive, which was how I did all my listening.

Part 3 of this series will cover grille effect and final testing, guides on designing your own enclosure, continued work on the second pair of boxes, and discussion of the system's sonic quality. R. O. Wright and Ed Dell will also describe their experiences listening to these systems.

TABLE 5

PARTS LIST FOR BOTH CROSSOVER BOARDS

(All capacitors are plastic-film units of 100V or greater.)

UNIT	VALUE	HOW PART WAS IMPLEMENTED
C1	43.8µF	parallel 15 + 15 + 10 + selected cap
C2	6.64µF	parallel 5.6 + .47 + .47 (selected)
C3	54.4µF	parallel 10 + 10 + 10 + 25 (selected)
C4	26.7µF	parallel 10 + 10 + 2.5 + selected cap
C5	11.1µF	parallel 10 + 1 + .47 (selected)
C6	96.4µF	parallel 25 + 25 + 25 + 10 + 10 + selected cap
C7	4.74µF	two small selected caps in parallel
C8	12.1µF	parallel 10 + 2.5 (selected)

Note: 25µF capacitors were purchased as 10µF and 15µF soldered in parallel.
(All inductors wound on ferrite [or powdered iron] bobbin cores.)

UNIT	VALUE	RDC Ω	WIRE	CORE HEIGHT & DIAM. IN INCHES
L1	1.64 mH	0.23	#18	0.99 × 1.38 (wound up 1.4)
A bigger core would be better.				
L2	0.239 mH	0.08	#18	1.08 × 1.13 (dewound 0.35)
L3	2.24 mH	0.27	#18	1.31 × 2.44 (dewound 3.0)
L4	0.911 mH	0.22	#20	1.07 × 1.11 (dewound 1.0)
This should have bigger wire and thus lower Rdc.				
L5	0.45 mH	0.10	#18	1.07 × 1.11 (dewound 0.5)
L6	1.08 mH	0.24	#20	1.07 × 1.11 (wound up 1.0)
L7	0.124 mH	0.09	#20	1.1 × 1.09 (dewound 0.35)
L8	0.178 mH	0.13	#22	0.73 × 0.86 (dewound 1.0)
TS2	6 terminal	Terminal barrier strip – #5 screws		
TS3	2 terminal	Terminal barrier strip – #5 screws		
TS4	2 terminal	Terminal barrier strip – #5 screws		
TS5	2 terminal	Terminal barrier strip – #5 screws		

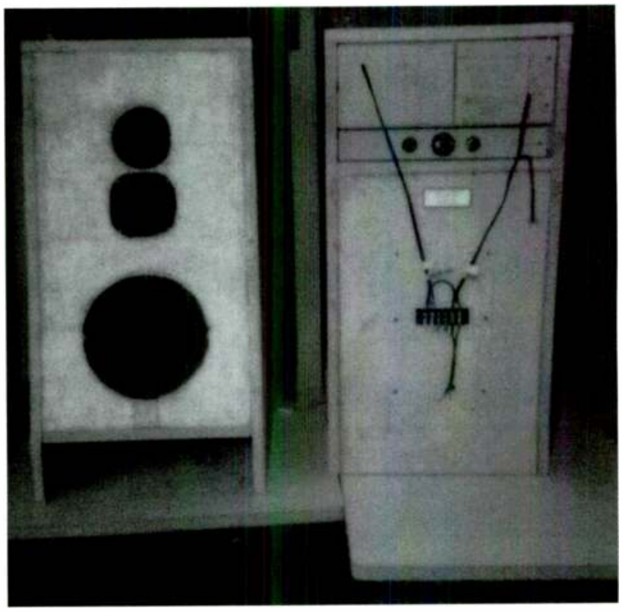


PHOTO 12: Front and back views of finished enclosures.

INDUCTORS FOR CROSSOVER NETWORKS

By Richard and Erin Honeycutt

A passive crossover network used to be considered a simple thing: just hook up a capacitor in series with the tweeter and an inductor in series with the woofer, and you have it. What size capacitor? Try 1/2πfR. What kind of capacitor? A nonpolar electrolytic. How about the coil? Oh, about a pound of #18 wire jumble-wound on a toilet-paper roller. Stick a bolt through the middle to mount it.

As my previous article "Caps for Passive Crossovers" (*SB* 3/92, p. 34) demonstrated, the traditional advice about capacitors was poor. You should begin by selecting the proper order of the filter. Then you choose whether or not to Zobel the speakers. Using software or formulae from *The Loudspeaker Design Cookbook*, you can determine component values. Finally, you select a polyester or polypropylene capacitor, or, if you must use 'lytics for financial reasons, you parallel them with polys.

The traditional advice about inductors was even worse than that for capacitors. To show you why, I must first review some basics.

INDUCTOR BEHAVIOR

Inductance is the property of a conductor that resists a change in current. If you connect a coil in series with a battery and a switch and then throw the switch, the current in the inductor does not immediately rise to maximum. Instead, the current exponentially approaches the maximum value, which you can find with Ohm's Law.

How long it takes the current to reach maximum is determined by the inductance of the coil and the resistance of the connecting wires, the switch contacts, and the battery itself. The more inductance, the more time delay. Inductance is measured in henries, and crossover inductors are usually in the millihenry range.

If you apply AC to the coil, the current tries to change all the time, and the inductor opposes that change. An inductor's opposition to the flow of AC is called the inductive reactance, X_L, and this depends upon the value of the inductance and the frequency of the AC. (At higher frequencies, the current is trying to change faster, so the inductor offers more opposition.) The formula is $X_L = 2\pi fL$.

Any piece of wire has inductance. When you wind the wire into a coil, the inductance increases, with the final value being approximately proportional to the square of the number of turns in the coil. You can reduce the number of turns necessary to yield a given inductance by inserting a magnetic core into the center of the coil.

A perfect inductor would have inductive reactance, but no resistance. How closely an inductor approaches no-resistance behavior is described by its quality factor, or Q. (The concept is the same as speaker Q: a ratio of power stored to power dissipated under certain conditions.)

The Q of a coil is given by: $Q = X_L/R$. Since a magnetic core reduces the number of turns needed for a coil of a given inductance, adding the core also increases the Q, all other things being equal. All other things, however, are not equal. To see why, you must look at the magnetic behavior of various materials.

PERMEABILITY FACTORS

Based on their behavior in a magnetic field, materials fall into three groups. When inserted into a magnetic field, some materials, called diamagnetic, very slightly weaken the field. A second group of materials, called paramagnetic, will very slightly strengthen a magnetic field. Finally, the third kind, ferromagnetic materials, will *greatly* strengthen the magnetic field.

The factor by which a given material changes the field is called its permeability. A material having a permeability of 1 would have no effect on a magnetic field—it would be truly nonmagnetic. Actually, there is no such material, but diamagnetic materials such as copper (permeability = 0.999) and

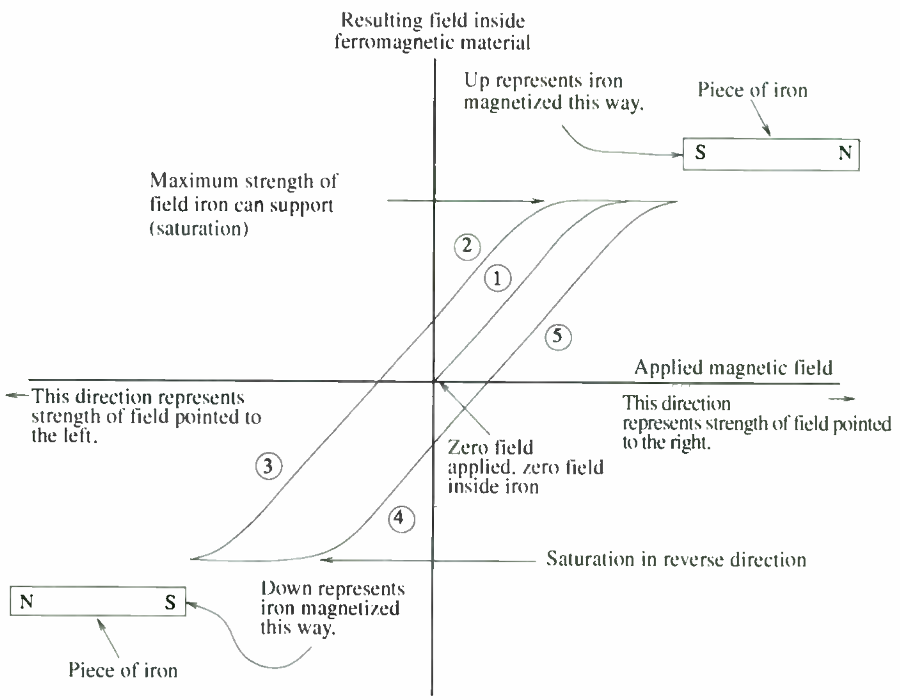


FIGURE 1: The hysteresis loop for a ferromagnetic material.

ABOUT THE AUTHOR

Richard Honeycutt was assisted on this article by his son, Erin, 19, a physics major at Wake Forest University, who has been assisting with sound-system installations since he was eight and helping to build speaker systems for commercial installation since the age of 13. Erin is also an amateur radio operator whose interests include music, basketball, and one special young lady.

FIGURE 2: Hysteresis loops for soft iron (left) and hard iron (right).

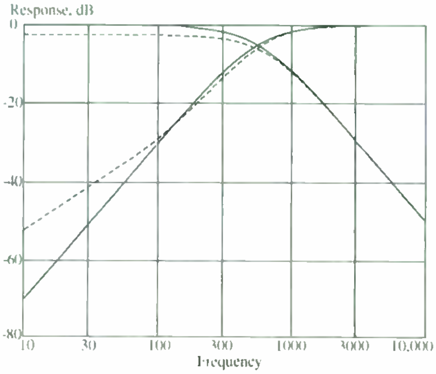


FIGURE 3: Comparison of second-order crossovers with low- and high-Q coils; (—— filter with high-Q coils; – – – filter with low-Q coils).

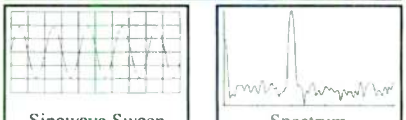
paramagnetic materials such as air (permeability = 1.0000004) come so close that they are often called nonmagnetic.

The ferromagnetic materials, on the other hand, have permeabilities significantly greater than 1. Iron, for example, is in the 200–1000 range, nickel is about 100, and Mumetal (an alloy of nickel, chromium, copper, and iron) is close to 20,000. There are also a number of ferromagnetic ceramics, called ferrites, that have significant permeabilities.

HYSTERESIS LOOP

A material's permeability is not constant, but depends upon the strength of the field in which it is placed. *Figure 1* shows a graph of the magnetic flux density inside a material plotted against the applied external field. Such a graph is called a hysteresis loop. The explanation of the curves is as follows:

1. The iron is initially unmagnetized. As the magnetic field is applied and increased, the field inside the iron increases to its maximum possible level.
2. The iron holds some magnetism as the field is reduced, even to zero.
3. The applied field's polarity is reversed, but some reverse field is needed to bring the field inside the iron down to zero. As the strength of the reversed field increases, the iron becomes magnetized in the reverse direction. The field inside the iron increases



until the maximum value is reached with the reversed polarity.

4. As the applied field is reduced, the field strength inside the iron decreases, but the iron still has some internal field even when the applied field is zero.

5. If the external field is now applied again with the original polarity, then increased, the field inside the iron will also be returned to its original polarity and increased.

There are two features of particular interest about a hysteresis loop. One is that above a certain applied field level, the field inside the material no longer increases. This condition is called magnetic saturation. In a coil core, the applied field results from current in the windings. Thus there is a maximum current that can be applied before saturation occurs. At saturation, the field inside the core, and thus the inductance of the coil, becomes a nonlinear function of the coil current. The result is distortion.

MAGNETIC MEMORY

The other feature is that reducing the applied field does not cause the internal field to drop to zero. This "magnetic memory" is what makes permanent magnets possible. A hard-iron material that would make a good permanent magnet would have a fairly square hysteresis loop, meaning that the material could be magnetized to saturation, and would stay magnetized almost at that level even when the external field was removed. Soft-iron materials used in coil and transformer cores have skinny hysteresis loops, meaning that they do not retain much magnetism, but more nearly follow the intensity of the applied field. *Figure 2* illustrates this difference.

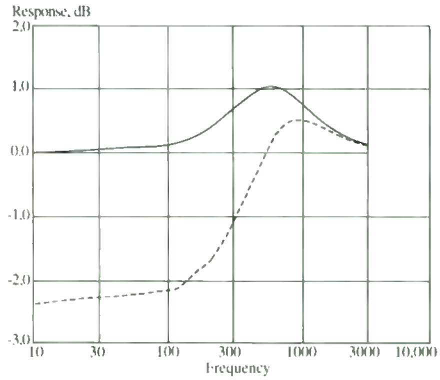


FIGURE 4: Composite response of second-order crossovers using low- and high-Q coils; (—— filter with high-Q coils; – – – filter with low-Q coils).

Soft-iron cores have skinny hysteresis loops and high saturation levels. Also, they enter saturation gradually. Unfortunately, iron is conductive, so that the alternating magnetic field caused by AC in the coil induces circulating currents in the core material. These "eddy" currents cause the core to heat up, and whenever heat is produced in an electric circuit, an equivalent resistance is present somewhere. In this case, the heating effect of the eddy currents appears as a resistance in parallel with the inductance of the coil, reducing the Q.

Breaking up the conducting mass into small regions reduces the eddy currents. Therefore, transformer cores are made of thin pieces of iron laminated together with an insulating varnish. Going one more step, you can further reduce the eddy currents by grinding the iron into filings and gluing them together with insulating epoxy.

Finally, you can use ferrite cores. Ferrites, being ceramic material rather than metal, are nonconductive and hence have no eddy currents. There are two problems with ferrites, however. The first is that you need a larger core than iron requires in order to avoid saturation. The second is that ferrites saturate more abruptly, causing more severe distortion.

AIR-CORE COILS

On the other hand, air-core coils do not have eddy-current problems, nor do they saturate. But since they require many more turns of wire, the wire's resistance lowers the Q. *Figure 3* shows the performance of a second-order crossover made, first, with a nearly ideal coil (Q = 1256), and second, with a low-Q coil (Q = 5, typical of a high-inductance air-core coil). Attenuation of a 20Hz signal to the tweeter is reduced from 58dB with the high-Q coil to 45.4dB with the low-Q one.

You can see the effect upon response and crossover frequency more clearly in *Fig. 4*. The low-Q coil produces a 2.3dB flat loss in the woofer output, and moves the crossover frequency up 6Hz. You must also consider the cost of the additional copper wire required by the air-core coil. So which kind of coil is best?

THE TESTS

I tested seven different 4mH coils in various ways, resulting in nine different configurations. The coils were:

- An air-core coil wound with 16-gauge wire and rated by the seller at 500W;
- Two 2mH 14-gauge air-core coils in series, each rated at 800W;
- A 16-gauge equivalent copper-foil air-core coil rated at 350W;
- An 18-gauge iron bar-core coil rated at 250W;
- An 18-gauge laminated-iron bar-core coil rated at 250W;
- A 16-gauge ferrite-core spooled coil rated at 500W;
- A 16-gauge ferrite-core toroid with a $1cm^2$ core cross-section, power not rated.

TABLE 1

MEASUREMENTS OF INDUCTANCE AND Q OF VARIOUS COILS

COIL	L (mH) (120Hz)	L (mH) (1kHz)	Q (120Hz)	Q (1kHz)
16-gauge air-core, original	4.71	3.99	4.72	4.65
16-gauge air-core, brass screw	4.72	3.99	4.74	4.65
16-gauge air-core, steel screw	4.89	4.01	4.57	4.44
16-gauge air-core, modified*	2.4	2.41	3.01	18.1
Two 14-gauge, air-core	4.07	4.04	4.2	15.6
Two 14-gauge, brass screw	4.07	4.04	4.2	15.6
Two 14-gauge, steel screw	4.41	4.27	4.2	14.3
copper foil, air-core **	4.03	3.91	3.46	16.1
18-gauge iron-bar-core	3.99	3.99	4.25	30.3
18-gauge laminated-iron bar core	4.05	3.9	7.25	21.7
16-gauge ferrite spool	3.93	3.91	13.16	71.4
19-gauge toroid	4.01	3.98	10.2	71.4

* See text
** Actually rated at 3.9mH

In addition, I tested two of the air-core coils with a brass mounting screw and then with a steel mounting screw in the center of the coil.

INDUCTANCE AND Q

The first test measured the inductance and Q of each coil at 120Hz and at 1kHz, using a Hewlett-Packard 4261A LCR meter (*Table 1*).

The inductance of a perfect coil would not change with frequency. Real coils, however, have distributed interwinding capacitance that causes the inductance to decrease somewhat as frequency increases. In fact, a real coil will have a self-resonant frequency at which the interwinding capacitance resonates with the inductance. Above this frequency, the coil is a capacitor. (For all the coils tested, the self-resonant frequency is well above the 50kHz limit of measurement.) Also, the core losses in a magnetic-cored coil will cause the inductance to decrease at higher frequencies.

The first and fifth rows in the table reveal a curious thing: although the 14-gauge coil has nearly constant inductance (within 0.75%) from 120Hz–1kHz, the 16-gauge coil does not. Since the coils were constructed nearly identically, I looked more closely

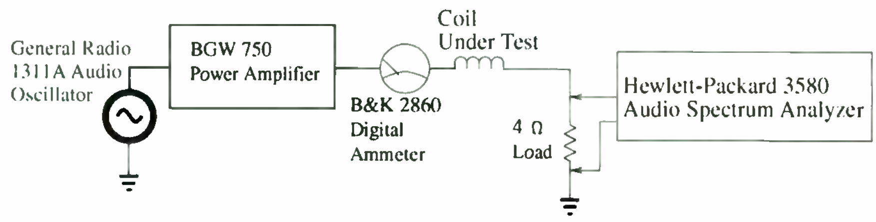


FIGURE 5: Test setup for distortion measurements.

and discovered that the 16-gauge coil was actually wound on a square iron tube, so that it was not an air-core coil at all, even though it was advertised as one! When I removed the tube, the inductance dropped by some 40%, but it no longer varied significantly with frequency.

Mounting the coils by means of a brass screw through the middle would provide a small amount of eddy-current damping, but this was too small to measure. Apparently this mounting method is acceptable. A steel screw, however, significantly increased the inductance of the coils, as you would expect.

The inductance of the copper-foil air-core coil varied more with frequency than did the traditional round-wire coils. This variation—about 3%—could be marginally significant in some designs.

CONSTANT INDUCTANCE

Surprisingly, the iron-bar-core coil had the most constant inductance of any magnetic-cored coil. Very likely this was because of the method of construction. The core in this coil was exactly the same length as the winding, which does not provide very effective flux linkage of the core.

The laminated-bar core extended beyond the ends of the coil by about an inch. This would make the core more effective in reducing the number of turns needed, but would also make the core characteristics more noticeable when you compare inductances at different frequencies. Both of the ferrite-core coils had very stable inductances.

As mentioned earlier, Q depends directly upon X_L, which is proportional to frequency. Q depends inversely upon R, which depends upon frequency only to the extent that nonperfect cores are used. (At higher frequencies, skin effect causes Q to decrease, but this effect is not important at audio frequencies. At 20kHz, skin effect would make at most a 1dB difference in the performance of a crossover coil. Much malarkey is bandied about concerning skin effect by people who have never investigated it—or who hope the audio consumer has not.)

You would therefore expect the Q of a coil to increase with frequency, and it did for all of the coils except the original 16-gauge "air-core" coil. This lack of variation in Q was another clue that something was strange about this unit. Of the air-core coils, the 14-gauge combination had the highest Q at 120Hz, while the modified 16-gauge had a higher Q at 1kHz. A high Q is one of the reasons for using a magnetic core, and as theory predicts, the iron- and ferrite-core coils do offer improvements in this respect.

HARMONIC DISTORTION

Since magnetic-core coils provide higher Q than air-core coils, the next question is how much you pay for that improvement in terms of increased distortion. To make a distortionless magnetic-core coil, the core material must have a perfectly linear hysteresis curve, and there must be no "magnetic memory." In other words, the curve would have to be infinitely skinny. Also, the material would have to be operated well below its saturation magnetization.

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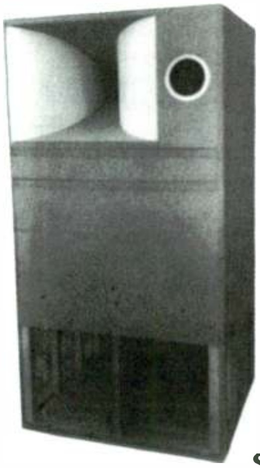

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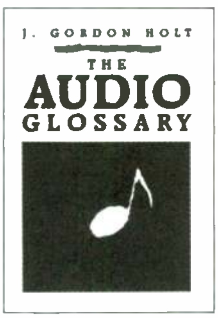

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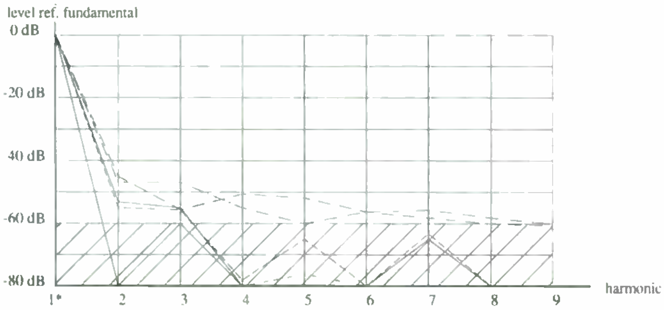

FIGURE 6: Distortion measurements for air-core coils; (—— 16-ga. air-core coil, original**; — — two 14-gauge coils in series; ---16-gauge coil with steel screw; —· — 14-gauge coils with steel screws; ——· 16-gauge air core, modified**; – —– –· copper-foil, air-core). Note: Shaded area represents equipment noise/distortion floor. *–fundamental; **–see text.

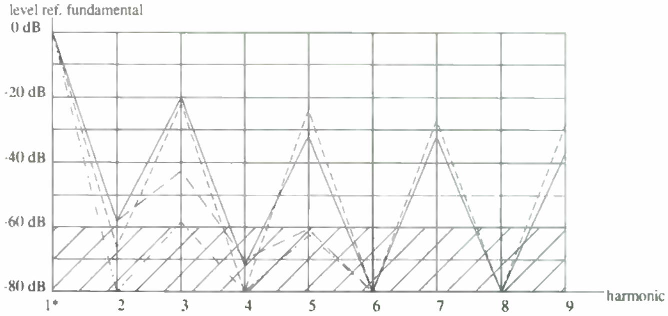

FIGURE 7: Distortion measurements for ferromagnetic-core coils; (—— ferrite bobbin core; — — iron bar-core coil; – – – ferrite toroid coil; –·–– laminated bar-core). Note: Shaded area represents equipment noise/distortion floor. *–fundamental.

Real magnetic core materials do produce distortion. To see just how much, I used the test setup of *Fig. 5*. This allowed me to feed each coil with a known current at whatever frequency I chose, and examine the output for distortion products. For the first test, I applied a 7A sine wave to the dummy load resistor, and found the resulting harmonic levels to be below –60dB, or 0.1%.

Then I applied the 7A sine wave to each coil; the results are graphed in *Fig. 6* for the air-core coils, and in *Fig. 7* for the magnetic-core coils. You might argue for varying the applied current according to the power rating of each coil. However, none of the power ratings included a speaker-impedance specification, nor is the current in a crossover coil the same among different coils in different orders of crossovers, even with a given speaker. Thus, using a standard current for all tests seemed more fair. (The variation in distortion with current is examined later.)

STEEL EFFECTS

Notice in *Fig. 6* that only the coils mounted with steel screws had distortion components above the measurement threshold of the test equipment. In other words, as you would expect, air-core coils do not produce significant distortion, although a small amount of distortion (about 0.1–0.5 %) is added if the coils are mounted using steel screws.

In *Fig. 7*, you see that the story is very different for ferromagnetic-core coils. The only one of these that had really low distortion was that using a laminated-iron core. Although this coil was rated at only 250W, it introduced no significant distortion. Second-best was the iron-bar-core coil, which intro-

duced about 0.16% second harmonic and 0.7% third harmonic. The ferrite-core coils were awful, providing about 10% third-harmonic distortion.

Defenders of truth and justice will be quick to point out that the test was unfair to ferromagnetic-core coils, since these are known to have a saturation point, and the 7A test current would represent almost 400W into an 8Ω load. Therefore, I measured the distortion in these coils again, this time varying the test current. The results are plotted in *Fig. 8*.

Notice that the ferrite-core toroid's distortion was off the scale, and the other ferrite-core coil showed a distortion of 5% just below 3A, corresponding to 72W into 8Ω. The 1% distortion level for that coil occurred well below 2A (36W). (Notice that the equivalent power figures would be only half as large if a 4Ω load were used.)

DISTORTION VS. FREQUENCY

Since there is some interplay between distortion levels and eddy currents, and eddy currents are frequency-dependent, I measured the distortion versus frequency. The results are shown in *Fig. 9*.

As you can see, frequency did affect THD differently for the three coils shown. As before, the distortion levels for the air-core coils were below the measurement floor, and those for the ferrite-core coils were off the top of the scale.

Harmonic distortion is not the only kind, and certainly is not the most objectionable. Intermodulation (IM) distortion results whenever two or more signals are applied to a nonlinear system. The result is that sum and difference frequencies are produced. Since these frequencies are not harmonically related to the signal, they can sound more or less like out-of-tune instruments. IM between two closely-spaced tones imparts a warbling sound. Either way, the effect is quite unpleasant.

IM distortion is audible at levels far lower than those at which you perceive harmonic distortion. I tested the IM distortion of the crossover coils by applying two tones to each: a 50Hz tone at 4.4A and an 80Hz at 10dB lower. I used the spectrum analyzer to measure the 130Hz sum and 30Hz difference frequencies. *Table 2* shows the results.

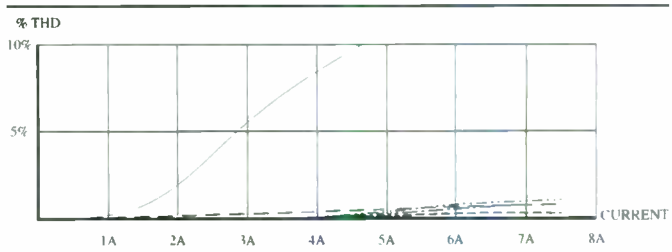

FIGURE 8: Distortion of coils vs. current (—— ferrite-core coil; — — iron bar-core coil; – – – 16-gauge air-core coil original; –·– two 14-gauge air-core coils in series; –··– 14-gauge coils with steel screws). Note: toroid distortion was >10% even at 1.5A. Coils not shown had distortion <0.1% at all current levels.

Notice again that the ferrite-core coils were the worst, with IM levels above 5%. The iron-core coils had IM below 0.5%, and the air-core coils could be coaxed into producing IM distortion only if a steel screw was used to mount them.

CONCLUSIONS

The coils tested had nominal values of 4mH, a value I chose so I could purchase units of the same inductance in as wide a variety of construction types as possible. Low-inductance coils are generally available solely as air-core types, and you can obtain very large ones only in the iron-core variety.

You would typically use the 4mH value

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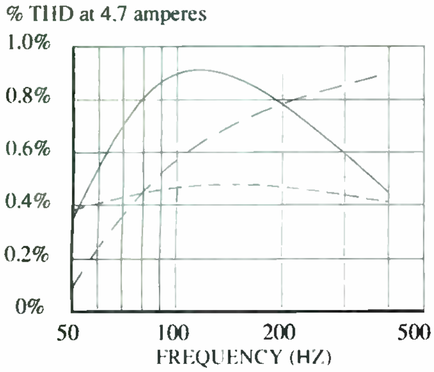

FIGURE 9: Distortion of coils vs. frequency; (—— iron bar-core coil; — — 16-gauge air-core coil with steel screw; – – – 14-gauge air-core coils with two steel screws).

in crossovers operating in the low hundreds of Hz, which was also appropriate, given the 120Hz and 1kHz test frequencies for measuring the inductance and Q with the H-P

TABLE 2

IM DISTORTION OF SELECTED COILS

COIL	% IM DISTORTION
16-gauge air-core, with steel screw	0.451%
Two 14-gauge air-core/steel screw	0.287%
Two 14-gauge air-core/2 steel screws	0.321%
18-gauge iron-bar-core	0.257%
16-gauge ferrite spool	6.34%
19-gauge toroid	5.05%

bridge. But, you'd expect the mechanisms for producing the distortion to affect lower- or higher-inductance coils similarly.

The conclusions seem pretty clear: ferrite-core coils are not suitable for crossovers. Air-core coils have vanishingly low distortion, but their Q is nothing to brag about. Iron-core coils behave somewhere between the other two categories, with laminated iron-bar cores perhaps the best compromise between low distortion and high Q.

Finally, the expensive copper-foil air-core coil, while offering by far the best Q of the air-core group at 1kHz, actually had a lower Q at 120Hz. So there is no ideal coil for crossovers. But by using the results given in this report judiciously, maybe you can reduce yet another source of distortion in your speaker systems.

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Ask SB

SPEAKER PLACEMENT

Although not many things in life are rocket science, I believe speaker building is. Once you consider all the factors and the problems to be overcome, it's amazing speakers sound as good as they do. One of the major problems is the room effect on the sound. Bill Waslo's article on focused arrays and minimizing room effects ("Focused Arrays: Minimizing Room Effects," *SB* 4/95, p. 10) was very interesting and provoked me to write.

D'Appolito configurations appear to be the way to go for numerous reasons. I have seen polar-axis response graphs for systems in which the interdriver spacing is two-thirds wavelength, but many times this spacing is difficult to achieve due to a high crossover point and large tweeter size (some almost 5" in diameter). What do the response graphs look like when the distance is greater than one wavelength? Also, what if the phase difference between the woofers and tweeter(s) is 45° rather than 0° and 90°?

Many audio/video speakers, especially THX certified systems, have two or three tweeters between the woofers, 3k crossover frequencies, and obviously don't meet the basic D'Appolito requirements. Their stated purpose is to narrow the vertical response, although some people contend that this causes them to have inferior room response and sound less musical.

I wonder whether the reason that some favor planar and ribbon speakers is because of their restricted vertical, and maybe horizontal, response, as well as their random reflections from the floor, ceiling, and side walls. We could build small speakers, and sit close, to minimize many problems.

I still haven't recovered from the first time I saw the Accuton Reference System One line array by Joe D'Appolito (*Fig. 1*). This masterpiece of audio indulgence included three Accuton tweeters, four Accuton midranges, and six Focal 5K013L woofers stacked from floor to ceiling in a three-way configuration.

In light of Mr. Waslo's comments, it would appear that such a system could benefit from superior vertical dispersion, a large sound-wave launch area (different issue), and random room reflections. Leaning the cabinets would increase the randomness of the reflections, but might prove difficult, as would the vertical off-axis response of such a system.

Robb Hayes
Newcastle, OK

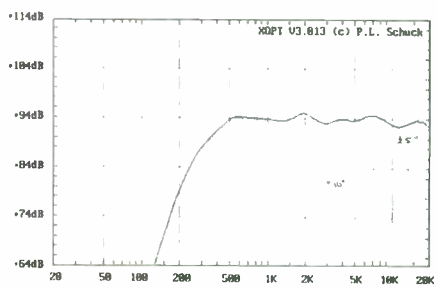

FIGURE 2: Midrange/tweeter response plots.

Contributing Editor Joe D'Appolito responds:

There are so many drivers in this system that I had to design it in two stages with the old software: LF/midrange and midrange/tweeter. Figure 2 *is a plot of the response of the midrange/tweeter combination (four C277s and the three C211s). The solid line is the on-axis response (relative to the center tweeter), while the dotted lines show the responses in the vertical direction at +5° and +10° off-axis.*

First, we see that the response degrades rather rapidly off-axis. Along the +5° axes, the spread of the tweeter array causes the high-frequency rolloff. The tweeter array produces a sharp null at about 9kHz along the +10° lines, while the four spaced midrange units produce a broader null around 2.5kHz. These response curves do not change very much for horizontal angles of +30°.

This shows that long line arrays tend to focus all the acoustic energy

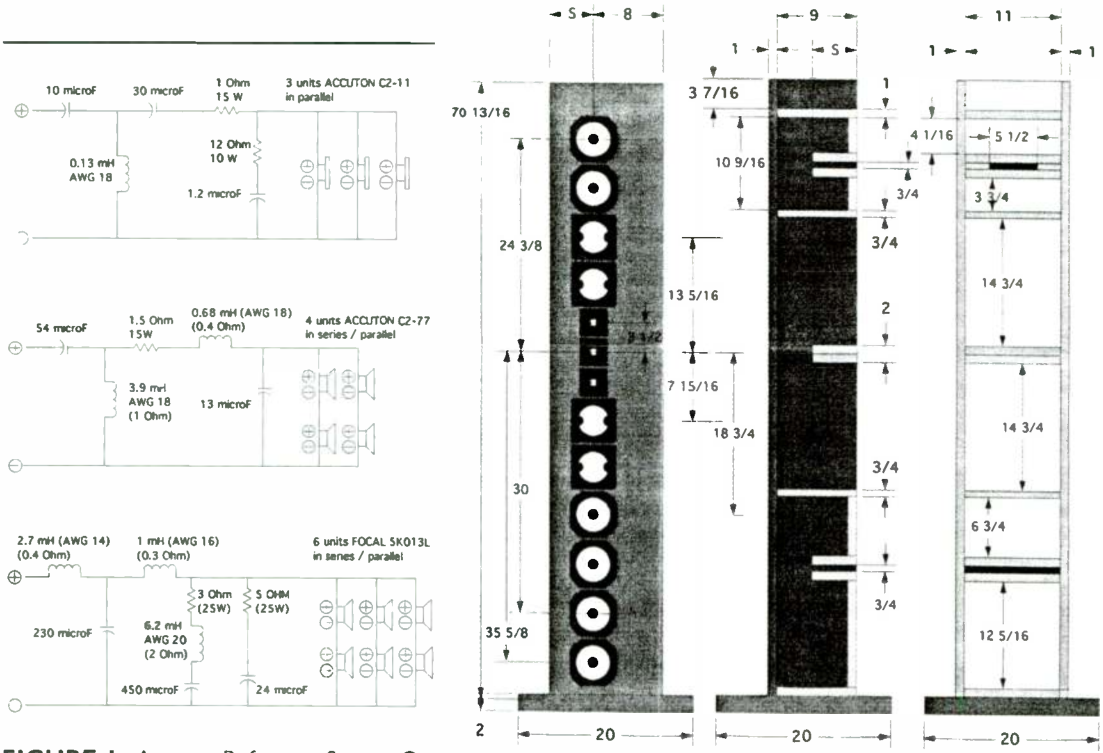

FIGURE 1: Accuton Reference System One.

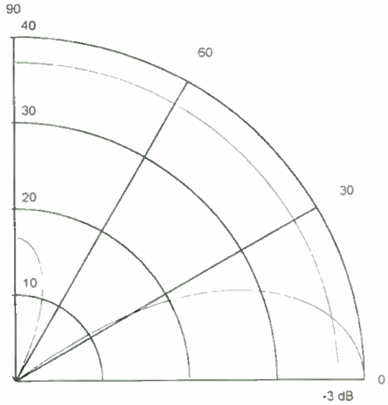

FIGURE 3: Vertical polar response of MTM array at two-thirds wavelength spacing, phi = 0°.

forward into a narrow vertical range. The center tweeter was set at typical seated listening height. The curves indicate that, for best response, the listener should be within the +5° envelope.

For the MTM array, Figs. 3–10 *show a sequence of vertical polar-response plots for various interdriver spacings and phase angles (phi). All plots assume point-source drivers. The effect of driver polar response will be limited to vertical angles of 45° or more for typical crossover frequencies and mid-bass drivers of 6″ diameter or less.*

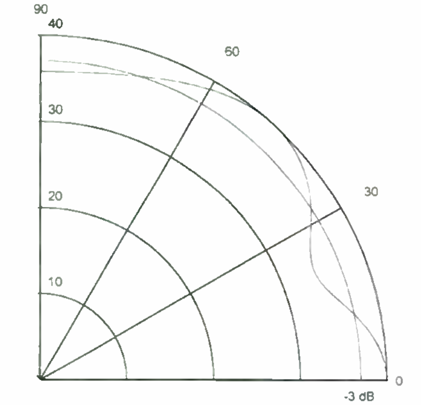

FIGURE 4: Phi = 90°.

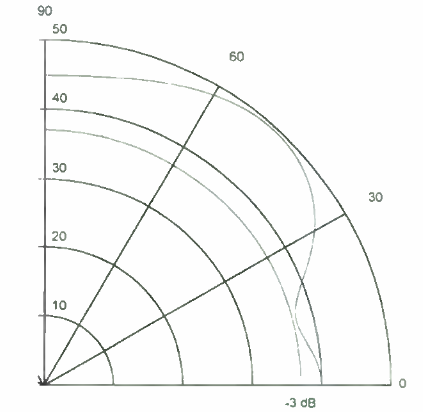

FIGURE 5: Phi = 120°.

Figures 3–6 *are for the different phase angles at the two-thirds wavelength spacing.* Figure 3 *shows the results for phi = 0. Notice that the array produces a strong forward lobe with a –3dB width of about +8°. With a sharp null at 50° and a greatly attenuated vertical lobe, this is rather ideal for most home environments.*

In Fig. 4, *phi = 90° which defocuses the array. There are no nulls, only a rippling of the response. (The MTM array is often placed horizontally when used as a center channel speaker in home theater applications.* Figure 4 *suggests that odd-order (90°) networks in this application will achieve the best horizontal dispersion.)*

Now for the surprise! Figure 5 *shows the polar response for phi = 120°, which is the phase angle between drivers using Small's parallel sum-to-one network. Notice that the off-axis response peaks almost 10dB at +45°! You should not use interdriver phase angles greater than 90° with the MTM array.*

I know of no crossover network that produces the interdriver 45° phase angle, but if there were one, it would look like Fig. 6.

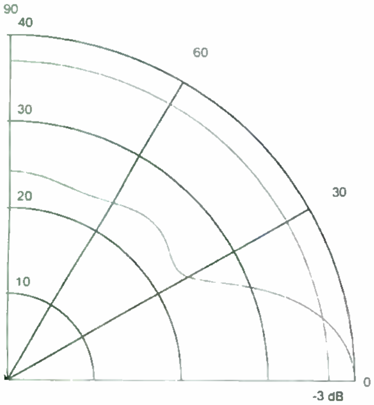

FIGURE 6: Phi = 45°.

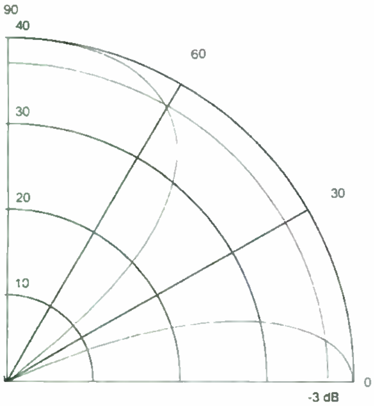

FIGURE 7: One wavelength spacing, phi = 0°.

For the effect of interdriver spacing, refer to Fig. 7, *which shows the response for phi = 0 and D = one wavelength. The off-axis null has moved down to 30°, but worse is the major off-axis lobe at 90°, which is as strong as the on-axis response. At 90° off-axis the drivers are exactly one wavelength apart and therefore add directly. At D = 1.5 wavelengths* (Fig. 8)*, the major off-axis lobe has moved down to +40° off-axis.*

When phi = 90°, the effect of interdriver spacing is not as critical. The array is still defocused. The major effect of increased spacing is to put more ripples in the polar response (Figs. 9 *and* 10)*.*

Incidentally, Morel (414 Harvard St., Brookline, MA 02146, 617-277-6663, FAX 617-277-2415) has a new line of compact soft dome tweeters that are only a little over 2" square. These tweeters seem ideal for the MTM array, as they allow much closer placement of the mid-bass drivers and can be crossed over at relatively low frequencies.

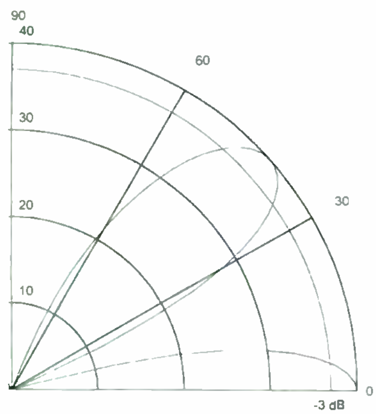

FIGURE 8: One-and-a-half wavelength spacing, phi = 0°.

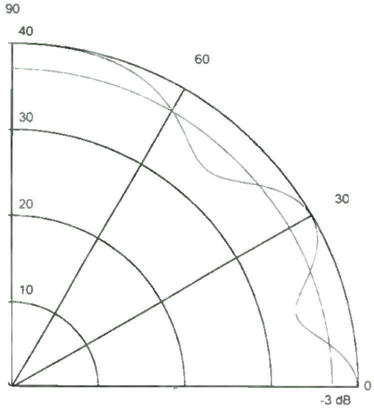

FIGURE 9: One wavelength spacing, phi = 0°.

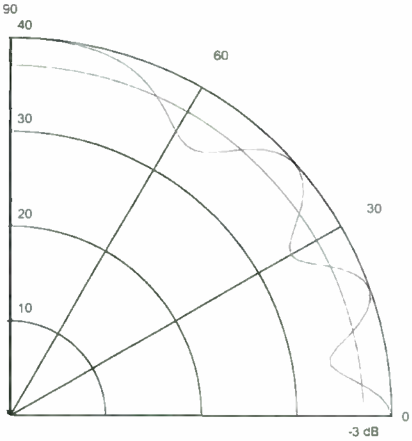

FIGURE 10: One-and-a-half wavelength spacing, phi = 90°.

CEPSTRUM CORRECTION

The graphic denoted as Figure 3 in Bill Waslo's "Reflecting on Echoes and the Cepstrum" (*SB* 5/96, p. 20) was incorrect; rather, the figure should have appeared as follows:

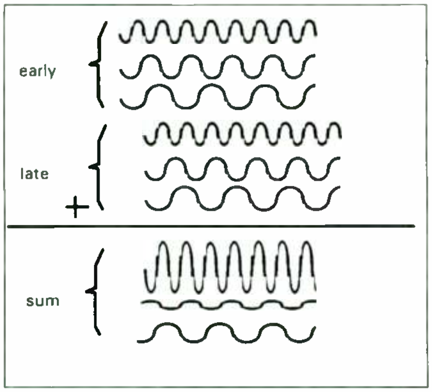

FIGURE 3: Sum of delayed sine-wave packets.

ELUSIVE TL DESIGN

David Field's article, "T-Rex: A Quarter-Wavelength Reflex Transmission Line" (*SB* 4/96), contains a good deal of common sense and touches on some of the many sore points of TL design, which nag those who have spent even a little time with these beasts. Part of the problem, as Field points out, is the chameleon-like nature of TL systems. By varying the length of the line, the cross-sectional area, the amount and type of stuffing, and the placement and number of chambers, you can force the system to behave like a bass-reflex box, a sealed box, a variously damped acoustic wave-guide, an aperiodic system, or practically anything else you want!

The only point in Field's excellent article that is a bit confusing concerns the stuffing material's influence in the line, which decreases the resonant mode peaks' amplitude and shifts their frequency. The shift in frequency is downward and occurs because the stuffing slows the speed of sound within the line; but it does so depending upon frequency. In other words, the lower the frequency, the greater the effect. Any calculations, and expectations, for a quarter-wave TL are valid only with an empty line and at one specific frequency.

The driver Q_{ts} is another variable. Old-line TL designers still prefer high-Q drivers, while Field advocates low-Q units. Alex Thornhill and I, however, obtained our best results with mid-Q woofers. I suspect good results are obtainable across a wide range of driver types if the units are matched correctly to the line. The plethora of component variables makes TL design a constantly moving target, and keeps the TL a bastion of "cut-and-try" speaker design.

I am surprised at the lack of attention paid to a very interesting piece of TL research done by Juha Backman from the acoustics laboratory at the University of Technology in Helsinki, Finland. Backman's paper, translated as "A Computational Model of Transmission Line Loudspeakers," was presented at the 92nd AES Convention in Vienna, March 1992, and is available as Preprint #3326 by phoning (212) 661-2355. The paper looks at many TL configurations—from the traditional TL of A.R. Bailey to modern, dual-ported designs—and presents informative graphs showing the effects of stuffing density and fiber diameter. It is a "must-read" for the serious TL enthusiast.

Robert J. Spear
Accokeek, MD

ACOUSTIC-MASS EFFECTS

Joe D'Appolito, answering Jay Doherty as to why *Boxmodel* and *Topbox* give different results for the same driver ("Mailbox," *SB* 5/96), brought up acoustic-mass loading as it affects the driver parameters used in design. I wish to echo Joe's response, while including additional comments.

Acoustic-mass loading seems to be understood by only a minority of speaker designers, including some who write enclosure-design programs. I read about this subject, originally, in N. W. Maclachlan's book, *Loudspeakers*—published by Dover, but now out of print. Maclachlan derives the concept of acoustic-mass loading from principles using solutions to the acoustic-wave equation, but also provides an intuitive justification for the concept.

You could imagine measuring the resonant frequency of a loudspeaker in outer space, where there is no air, no acoustic waves, and no acoustic-mass loading. This measurement is the mechanical resonant frequency. Back in the atmosphere, you can measure the resonant frequency with the speaker mounted in the middle of an infinitely long pipe having an inside diameter the same as the speaker's active diameter. According to the wave equation, the speaker will have the same resonant frequency in the latter case as it did in outer space.

On the other hand, when measured in free air—or in an open or sealed enclosure—the speaker's mass will have increased after compensating for stiffening effects of air trapped in the enclosure. The explanation is that acoustic-mass loading is caused by the velocity of the air particles in a wave, in which a component is not parallel to the direction the diaphragm is moving. In other words, whenever air particles can move sideways relative to the diaphragm's motion, the acoustic mass will increase.

The acoustic-mass loading, which must be added to the mechanical mass of the diaphragm to find its total mass, must be known and added in for each condition of measurement—in free air, or in an enclosure. The value in an enclosure depends on its size and whether you consider the outer or inner surface of the diaphragm. Suitable equations for these calculations are given in *Acoustics* by Leo L. Beranek (available from Old Colony Sound Lab as #BKAC5 for $44.95, plus S/H).

Dr. Victor Staggs
Orange, CA

CAPACITOR CONFLICT

I would like to reply to Mr. Moncrieff regarding his response to my letter (*SB* 2/96, p. 51) about his claims for the InfiniCap (that other caps "smear music by a million to one"). He said that I "succumbed to a common mistake (even among engineers), confusing two different kinds of delay and smearing." Well, Mr. Moncrieff, first of all, I am an engineer, and more importantly, I am well aware of the smearing type of delay, known as non-constant group delay vs. frequency, or phase distortion.

Years ago I presented a paper at the Boston Audio Society on the audibility of phase distortion, and more recently I did research on this with precision all-pass networks, listening to live-miked acoustic transients through headphones. Both experiments showed a maximum audibility when the group-delay variation was centered near 200Hz. But about 1ms of delay smearing was needed in order to barely hear the transient blurring, and the pulse distortion was clearly visible on a scope (unlike nonlinear distortion, the threshold of which eludes visual observation).

So, if Mr. Moncrieff would publish a scope photo of the alleged "infinite number of different-looking waveforms" distorted by a "conventional capacitor," or would send me an InfiniCap sample so I can observe the effect, then I will be happy to be "enlightened."

Mr. Moncrieff further claimed that a million-to-one inductance variation through various paths in a cap "could smear music all the way down to 2Hz," assuming a self-resonant frequency of 2MHz typical for "ordinary" caps around 1µF.

Now, he states that delay is proportional to L, but this is only true in an L-R (1st-order) circuit. But a wound capacitor is primarily a distributed L-C circuit, with very low resistance. The delay in an L-C network varies as √L, not L. So the 2MHz resonance with the assumed million-to-one inductance variation would have a possible effect only down to 2kHz, not 2Hz.

Let's look at some physics. The length in a wound cap is around 100′, which has an electrical delay of 100ns (one wavelength at 10MHz), while 2Hz has an (electrical) wavelength of 93,000 miles! For a 100′ maximum path length to influence 2Hz is analogous to hearing a ½-second echo from any acoustic structure whose maximum dimension is 0.0014″!

Sure, one could imagine 560′ of one-millionth-inch diameter tubing crammed into a dust particle, giving a ½-second acoustic delay. But the laws of physics have something to say about energy—how much of that echo would you hear (with 1000 to 1,000,000dB of loss in the tubing)?

Since my first letter, I have measured some components (*Fig. 1*). For example, I tested a relatively "ordinary" capacitor—a Siemens (I think, from its logo, although what I had ordered was a Chateauroux) part #PB1000MKP-FC 10µF polypropylene. The self-resonant frequency was 131kHz, the series inductance 148nH, and the series resistance 0.0175Ω. Although the resonance was lower than the 2MHz I mentioned (appropriate for smaller caps), the square-wave response, with either high- or low-pass configuration with a resistor, was as perfect an exponential decay as you could hope to see on the scope over the full audio range, with no variety of distorted waveforms.

A speaker I recently designed, using these caps, sounds excellent, as clear as the midrange alone with no crossover. Note that the inductance of that cap, 148nH, is only about that of a 1′ speaker cable. This is because even though the cap's wound length may be 100′, the two plates are so close that their magnetic induction nearly cancels, as in any good transmission line.

I also measured a "regular" wire-wound, sand-cast 3.3Ω 25W resistor, a Colher CW 25E ($0.60). Its inductance was 24nH, causing a 20kHz phaseshift of 0.05°. The impedance rose by 1dB at 11MHz. Now, in the catalog with this resistor is also a "noninductive" unit, for $1.80. Those who can hear above 11MHz should use this one! (Sorry, I couldn't "resist" that!)

By the way, how is it so easy to hear the

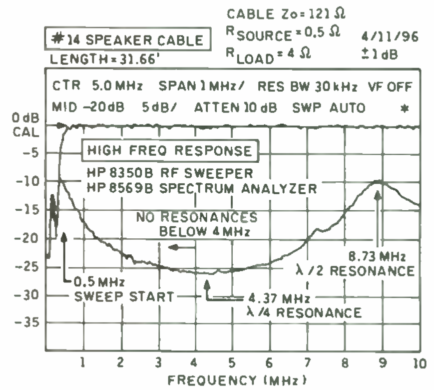

FIGURE 1: Reader Colin's component measurements.

"astounding" improvement of a cap when even the best recordings are made through microphone and other electronics, most of which probably don't have "audiophile" capacitors?

I don't mean to belittle quality. As an engineer and an avid musical-reproduction researcher, I certainly appreciate the admirable pursuit of perfection in Mr. Moncrieff's capacitors and other fine components. But regarding the astrophysical claims made in ads, would some non-market-vested entity with audio wisdom please conduct careful double-blind scientific comparisons, using ruthlessly perceptive and politically incorrect listeners, and publish the results?

Perhaps if *Consumer Reports* hired some audio engineers, the Boston Symphony, and a few Tibetan Buddhist monks who can hear the sound of one capacitor plate charging....

Dennis Colin
Barnstead, NH

J. Peter Moncrieff responds:

Two facts are indisputable. InfiniCap's design is different from other capacitors, and so is its sound (as confirmed by numerous independent evaluators). As a research scientist, I believe in cause and effect, not mysticism. Thus, the correct explanation for InfiniCap's different sound has to be its different design.

Most engineers are also open-minded scientists, but Mr. Colin's comments remind me of the letters one sees in other magazines, where a vociferous minority of engineers argue that different cables with the same LCR can't possibly sound different, that amplifiers with the same frequency response can't possibly sound different, or that we can't possibly hear differences when employing a signal that's already been corrupted by miles of poor cables, poor capacitors, phase rotations, and so on.

These engineers were taught a limited set of rules, which only partially describe a limited set of phenomena. From this circumscribed basis, they argue a priori *about what can't exist in the real world. That's a perversion of science and the scientific method that should have disappeared with the church elders telling Galileo what couldn't* *possibly be. There are more things in heaven and earth than are dreamt of in these engineers' philosophy, or in their limited set of engineering rules.*

A research scientist openly accepts the world as it really is, and tries to explain it. New theories may be needed to explain empirical evidence. We don't use old theories to argue a priori *that real empirical evidence can't exist. I (and other research scientists in audio) routinely evaluate audible differences and what causes them. We have discovered sonically significant factors and phenomena that would make these engineers' hair curl.*

A scientist's job is to learn by opening new paths in new directions. Maybe our training schools are to blame for teaching engineering rules as if they were horse's blinders—so some engineers blindly follow the old straight and narrow path, while zealously refuting the existence of new real-world phenomena all around them.

In his zeal, Mr. Colin even repeats in his second letter the same principal blunder I pointed out in his first letter—conflating two types of delay. He bases his argument on the propagation delay of a 2Hz wavelength, which he then paints as absurd. But propagation-delay smear in a conventional capacitor is only 1000:1, not 1,000,000:1—as pointed out in our ad and in my response to Mr. Colin's first letter (so 2Hz is an irrelevant straw man).

I'm happy to debate someone who's open minded, rational, and who listens and learns. But I wouldn't even hire as an engineering assistant someone who repeats the same principal blunder, or who sets up phony straw men to sarcastically knock down.

Mr. Colin and I have an interesting disagreement about the other kind of delay, caused by inductance smear, which is 1,000,000:1 in a conventional capacitor. He assumes that any adverse effect of this inductance would arise only from altering the capacitor's internal self-resonant frequency (a √L phenomenon), while I assume that it could also arise from the higher inductance paths interacting with the external circuit to which the capacitor is connected (this effect is proportional to L, not √L).

We each could make a case, depending on the circuit application of the capacitor. But Mr. Colin overlooks the fact that, even if I were to grant him this point completely, we'd still be looking at conventional capacitors smearing music above 2kHz (instead of above 2Hz), and there's plenty of music above 2kHz, including the spectral region where the ear is most sensitive.

Mr. Colin's best point is a challenge to measure the phenomenon of signal time smearing, to prove the nexus between InfiniCap's different design and its different sound, and to prove that this nexus is related to time smearing of a signal. As most readers know, Mr. Colin's sine-wave sweep LCR/Z capacitor test is irrelevant to measuring such time smear, so his a priori *speculations that this phenomenon can't exist or can't be heard are likewise irrelevant.*

Mr. Colin's test is irrelevant because only a single, simple sine wave is input at any one time, because the output is measured as an average amplitude over time, and because the output at any one instant is the sum of all the different paths through the capacitor arriving at that instant. Thus, this measurement can't see the time-smear effects of the different paths, it can't evaluate time-smearing effects upon a complex signal, and it can't observe instantaneous amplitude (so it can't even see time smear).

But there is a test that is relevant: take a snapshot of real music. Music provides a complex signal input, and the instantaneous snapshot can observe time-smearing phenomena. Real music is also the only test signal that really matters, for we use our capacitors to listen to music, not sine waves.

If the scope shows the same kind of time-smearing differences that are predictable from InfiniCap's different design, and the same kind of time-smearing differences that independent evaluators hear, then we as research scientists have a pretty ironclad case against any old-fashioned engineers who still say that the earth can't revolve around the sun.

We know that a trumpet's sound is very revealing of sonic differences in audio. This trumpet's waveform consists of positive spikes, each about 75µs wide. Within the overall waveform of each spike, there are numerous little peaks and valleys. These details give the trumpet sound its brassy texture; they make it sound real.

The left waveform (Fig. 2) *is the original trumpet waveform, measured through a straight wire from an M&K RealTime CD,*

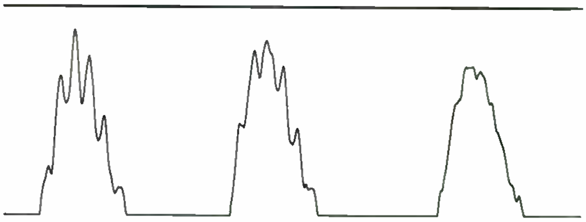

FIGURE 2: Waveform of a trumpet note (left). The same note measured through InfiniCap (center) and through a multiple capacitor (right).

on a 1MHz 16-bit sampling scope. Notice all the little peaks and valleys that make the trumpet sound real.

The center waveform is precisely the same instant of trumpet note, measured through InfiniCap. Notice that every one of the little peaks and valleys is still represented, and in reasonable approximation to the original. The brassy texture of the trumpet is preserved and transparently revealed, so the music sounds real.

The right waveform is precisely the same instant of trumpet note, measured through a multiple capacitor with ten coaxial sections (which is hypothetically better than a conventional capacitor with only one section). Notice that the subtle musical information represented by all those little peaks and valleys has been virtually wiped out. Some energy from each little peak has been delayed and smeared in time, diminishing that peak and filling in the little valley following. So the music signal has been time-smeared into an averaged waveform. This trumpet waveform has lost the subtle texture that makes it sound real.

AND, MOREOVER...

Referencing Dennis Colin's letter, and J. Peter Moncrieff's response ("Mailbox," *SB* 2/96, p. 51), I would also like to hear a credible explanation for the InfiniCap ads. These very ads are what I use as an example for what gives the high end a bad name.

The ad claims that small delays in "ordinary" capacitors are sonically significant. Dennis Colin correctly points out that these delays are at least 100 times smaller than those of conventional loudspeakers. In fact, the difference between the high-frequency phase shift of two otherwise-excellent loudspeakers is often 100 times that introduced by even the worst capacitors.

Mr. Moncrieff's argument about the "lagging waveforms corrupting the main (earliest) version" is nonsense. His idea is that a conventional capacitor divides the signal into many copies, and recombines them after delaying each copy by different amounts. This would be a problem if the delays differed by, say, −180°; but, as Mr. Colin notes, this application deals with delays on the order of 0.6° at 20kHz. Combining a 0.6° phase-shift signal with a "wildly different" one, having 0° phase shift, will not corrupt the signal by any significant amount.

Mr. Moncrieff claims that his customers can hear significant benefits from substituting InfiniCaps, which is what he calls "hard empirical evidence." It certainly is possible that the InfiniCaps' construction produces audible benefits. He fails, however, to address Mr. Colin's complaint against the advertisement: how can the delay caused by a capacitor be audibly significant when the delays already present in the audio chain—comprising microphone to recording medium, to loudspeaker, to ear—are much more so?

Other advertisements may be equally misleading, but the InfiniCap ads are especially onerous. They use such ambiguous and emotional phrases as "smears music by a factor of 1,000,000:1." I know where the 1,000,000:1 figure originates, but what does it mean to "smear music?"

Ralph Gonzalez
<gonzalez@crab.rutgers.edu>

J. Peter Moncrieff responds:

My previous response to Dennis Colin's second letter generally suffices to address Ralph Gonzalez's concerns here. It's worth noting that Mr. Gonzalez, like Mr. Colin, confines his "it's a small problem" critique to only one kind of delay, propagation delay. But our ad prominently states that inductance-delay smear is a worse problem than this propagation-delay smear.

Indeed, the various length propagation paths shown in our first diagram were conceived primarily as a pedagogical stepping stone. The reader could then visualize, using our second diagram, how different length paths have drastically disparate inductances, constituting a worse problem. Similarly, the sentence "this time smears your music, so it sounds..." was originally in the text which described the second and worse problem, but was moved to the first section to help readers grasp the concept of time smear, aided by the disparate paths shown in their simple linear form instead of their convoluted three-dimensional form.

Colin and Gonzalez ask for the dismissal of an entire advertisement based on the charge that it calls propagation delay a small problem. That's neither scientific nor fair, since the ad itself states that it is the lesser problem.

Ironically, Mr. Gonzalez—unlike Mr. Colin—says that he understands the derivation and concept of the 1,000,000:1 ratio, among disparate inductances and various paths through a conventional capacitor. Why does Mr. Gonzalez limit his critique to only propagation delay, the lesser problem? Surely, he must recognize the importance of the worse problem discussed in our ad: some paths, with a million times higher inductance, output a version of the music signal with a million times slower rise time than the version output by the shortest paths. If that isn't worthy of being called music smearing, what is?

STICKY SITUATION

I need to clean some speakers. I think soda spilled on the speakers and one of them seems stuck. The cone moves, but very slowly—as though there is sugar around it. Is there anything that I can do?

James Banfield
banfield@mainelink.net

Perhaps Martha Stewart is reading this and will offer a reply.—Eds.

HELP WANTED

[We encourage readers who may have information on the following topics to correspond directly with these letter-writers.—Eds.]

I have a pair of Infinity Quantum 5s. I like the speakers, but have had them for a long time. How can I upgrade the sound? Can the components of the crossover be replaced? What about tri-amping? Should I say good-bye to them and build replacements? If so, could you recommend a design; perhaps one from *SB*?

Steve Tatum
784 Dexter Drive
Broomfield, CO 80020

I want to rebuild a pair of JBL Century K100 speakers, a consumer version of the 4311 studio monitor, with serial numbers 115665A and 115666A and manufactured in approximately 1973. I would rewire the speakers, eliminating the presence and brilliance controls from the circuit; install new binding posts; and replace or update the crossover network. Does anyone have any advice on how to do it?

Stephen M. Middleton
PO Box 1511
Foley, AL 36536
(334) 971-1111, FAX (334) 970-1116

9Hz in a Barrel

from page 10

ten to. Reference Recordings' "Pomp & Pipes" RR-58CD is spectacular, but most of the music is not to my taste. Many Telarc recordings are also spectacular. A good sampling is "Organ Blasters" CD80277.

If you want an inexpensive and easy-to-build subwoofer that really extends the frequency response of your system to subaudible frequencies, this design will satisfy your needs.

References

1. D.B. Keele, "Low-Frequency Loudspeaker Assessment by Nearfield Sound-Pressure Measurement," *Loudspeaker Anthology, Vol. 1*, JAES 1980.

SOURCES

Speaker	Radio Shack 40-1301
55 gal. drum	Northern Hydraulics PO Box 1499, Burnsville, MN 55337 (800) 533-5545
BassBox 5.1	Harris Technologies PO Box 622, Edwardsburg, MI 49112 (616) 641-5924
Quick Box and Bandpass Boxmodel	Old Colony Sound Lab PO Box 243, Peterborough, NH 03458 (603) 924-6371

Classifieds

TRADE

STATE-OF-THE-ART PASSIVE CROSSOVERS featuring Hovland MusiCaps. Software available, free design guide. ***ALLPASS TECHNOLOGIES, INC.***, 2844 Charmont Dr., Apopka, FL 32703-5972, (407) 786-0623.

THE BEST MUSIC YOU MAY NEVER HEAR. Free catalog. ***CREATIVE MUSICIANS COALITION***, Dept. GBM, 1024 W. Wilcox Ave., Peoria, IL 61604, (800) 882-4262.

PAIR DYNACO MKIII, absolutely mint condition, $395; JVC Stereo Integrated Amp A-X400, excellent condition, $190; manuals included. Place orders and make payments to: **GÖSTA BERLING**, Box 132, Nakusp, BC V0G 1R0, CANADA, phone/FAX (604) 265-3175.

ELECTROSTATIC electronics, transformers, Mylar®, information. From author ***ROGER SANDERS***, (505) 759-3822.

NEW!! SILVER SONIC digital interconnect cables are now available, 75Ω coax & 110Ω AES/EBU balanced; state-of-the-art performance, affordable prices. For information, contact ***D.H. Labs***, PO Box 31598, Palm Beach Gardens, FL 33420-1598, phone/FAX (561) 625-8998.

FOR SALE

Pair Electro-Voice EVM-15L woofers, new, $165. Darren, (561) 625-8998.

Two Ongaku Audio Note power transformers; new, never been used; $175 each, asking $125. Two Audio Note chokes 10H 200mA, new $65, asking $40 each. Ashby, (954) 723-5225.

Heathkit IO-4540, 5MHz single-trace oscilloscope, asking $100 shipped. I would like to trade my Dynaco ST-70 with Van Alstine driver modification for a Bryston power amp. Greg Nawrocki, 21 Indiana St., Kitchener, ON N2H 2A4, CANADA, (519) 745-1579, gnawrock@ionline.net.

Bitwise Music Systems D/A, $995; Reference Line 1 MKII remote passive preamp, mint condition, $850; set of three aluminum cones, 1.5″ diameter × 1.5″ tall, bolts, $19; pair JVC HSW1101-01A ribbon-leaf tweeters, enclosures, crossovers, $50; pair Techniques EAS-10TH 400A ribbon-leaf tweeters, enclosures, crossovers, $95. Steve, (203) 397-3888 in Connecticut.

Liberty Instruments' Audiosuite v1.16, complete systems, ready to install and run. Includes Echo DSP soundcard, calibrated microphone, microphone preamp, all cables and probes, manual and software, mint condition, hardly used, six months old, $475. Upgradeable to v2.0 for $100. Henry Lugg, (510) 837-7817.

Seeking a good home: pair Peerless 1727 woofers, $100; pair Peerless 1858 woofers, $90; pair Focal 7N412 DBs, $50, (two pair with curves); Adcom 2535 (4 x 60W or 2 x 60W plus 1 x 200W, level controls); $360, all unused and in mint condition. Also, N.E.W. A-20.1 Class A amp, $450. Duke, (504) 486-7943.

Two pair Music Posts, $25; (R/L pairs) Audio Quest panel-mount RCA connectors, $8; Noble pots, two 20k and one 25k, $8 each; TI 486DX2-80 CPU, $15; video capture card, $45; VESA IDE cache controller card, $20; VESA graphics accelerator card, $20. Dan Patten, dpatten@dasengr.com, (801) 225-8577.

Pair JBL 2440 horn drivers (same as 375); pair JBL 2328 horn adaptors; pair Altec 291A drivers; pair Altec N-800-D crossovers; pair Altec 421 15″ woofers, pair Altec 808-8B drivers; pair EAW horns, fit 802/808 drivers; pair large Community horns fit 2440/41; Van Alstine Model I rack-mount solid-state preamp. David, (914) 688-5024.

WANTED

Owner's manual for EICO Scope model 425, original or copy. Call Dave collect, (518) 399-3798.

Altec 838 or 820, Laguna, Carmel, or any other comer speaker; 604, 605, 601 coaxials. Tannoy Red or Gold monitors; Jensen CX-222, G-600, G-610; other comer and coaxial/triaxial speakers. Sonny, (405) 737-3312, FAX (405) 737-3355.

Photocopy of schematic/service manual of 3M-Mincom Division 6100A audio test set. Single partridge OTF UL2. Willie, Toronto, CANADA, (416) 251-3007.

Ad Index

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

MODIFYING VIFAS

Since they first came out, I have been using the Vifa "silk-dome" tweeters, especially the D27TG-35-06, for the cheaper speakers I build. I consider them a very good value. However, they definitely don't have the clarity of more expensive models, such as the Dynaudio Esotec, which I have also used in several sets of speakers. Specifically, the Vifas aren't as clear as the Esotecs at any power level, and their sound "hardens up" much sooner as power levels increase.

MODIFYING THE VIFA

Even so, they are very good for the money, so I thought it might be worthwhile trying to modify them to see if I could at least narrow the gap. After trying a number of mods, I came up with one that is easy, simple, and quite effective.

A comparison of the Vifa with the Esotec shows that the most obvious physical difference is the structure that dissipates the backwave. The Esotec has a number of sophisticated design features to do this, but the Vifa does not (quite understandable at its price).

If you take the back off one of the Vifas, you find that the bore in the pole piece is empty, and the only measure taken to dissipate the back wave is a simple layer of felt in the back chamber. By contrast, the Esotec has a bore that is stuffed, along with a more sophisticated back-chamber shape.

I decided to see if adding stuffing to the Vifa would help its clarity. It did. Before-and-after SPL checks (I use a Mitey Mike) show that a couple of small peaks, which seem to be due to cavity resonance, almost vanish. Listening tests show a significant improvement in clarity, and less "hardening" at higher outputs. No, it's still not as good as an Esotec, but it's a lot closer.

THE PROCEDURE

I have modified about 20 tweeters now, at almost no cost, and have evolved the following simple procedure. (For those who worry about such things, yes, I'm sure it would void any warranty on the tweeters.) Just be careful to protect the dome while you're working on the tweeter. Beyond dome damage, though, I don't see how anything could easily go wrong.

Remove the back chamber, which is easiest to do by cutting into its plastic edge, just against the magnet, with a hacksaw blade. That provides a space that lets you pry off the chamber (use a screwdriver blade—gently!). The chamber comes off fairly easily, and there is nothing delicate you might damage in the immediate area. Next, simply place the tweeter face down on a clean table top. Since the dome is recessed, this position actually protects it from damage.

Now stuff the pole-piece bore. I use polyester batting, contained in a fine tulle netting, which is very cheap at any fabric store. The actual amount of stuffing is not critical. I have tried a wide range of stuffing densities. If you make it very dense, the tweeter resonance drops, but there's no need to go that far. I now put in just enough stuffing so it will stay securely in place, using the following steps:

- Roll a plug of the polyester approximately 1″ long, with a diameter larger than the bore, and center it in a patch of the netting about 3″ square.
- Shape the netting around the polyester into a small cylinder. Gently insert the end of the cylinder that is closed by the netting into the bore until you feel it pressing against the tweeter dome. (It's not difficult. The dome is quite strong from that direction. If you watch the dome as you are doing it, you can see it move outward very slightly.)
- Pull the plug back about 1/8″, then cut off any polyester that is still protruding from the bore. Leave the netting that protrudes, since it will act to stop the plug from moving, once the back is in place.
- Replace the back with a bit of glue to hold it in place (I use silicone seal).

That's it! I consider the mod very worthwhile, since it does make an easily audible difference for almost no cost.

Larry Van Wormer
Port Elgin, ON, Canada

Reader Service #19 →