DESIGNING & TESTING WITH ELECTRONICS WORKBENCH

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Speaker Builder

THE LOUDSPEAKER JOURNAL

BUILD A PAIR OF RIBBONS

PC DESIGNING A 3-WAY TL

TESTING THE WAVEGUIDE

THE PC MOVE ON MUSIC

FASTENERS GALORE

TESTDRIVING AUDIOCONTROL'S C-101 III ANALYZER

Critics Judge New InfiniCap

(bulletin posted on The Audiophile Network, to one and all)

Note: Mr. Blackburn's credentials as a sober critic are impressive. An engineer for a worldwide company, he is involved in the development and ongoing support of advanced digital imaging and storage systems. His articles have been published by Stereophile (he satirized pseudo-physics in high end audio), The Audiophile Voice (equipment and music reviews), and The Sensible Sound (music reviews). He is a regular contributor to Positive Feedback.

Msg#:19460 09-11-95 08:46:01 From: Doug Blackburn
To: All
Subj: New Caps

I found a great sounding new cap — unbelievable sounding actually. I used to think [a highly regarded multiple section film and foil capacitor*] sounded pretty good, but ***these*** caps are unreal. I tried these caps out in various locations in some of my equipment — power supply bypasses, in the audio signal path, etc. ***Unreal*** sound quality. These make [a highly regarded multiple section film and foil capacitor] sound ***broken!*** I'm ***not kidding!*** The difference is ***very*** large.

These are called InfiniCap. They are from the same people who came up with WonderCaps quite a few years back. These sound ***nothing*** like WonderCaps — far far far better. They aren't cheap, but they don't set new record high prices either.

If I'm reading the tech literature correctly, they sound so good because there are multiple spiral foils inside instead of a continuous foil. At the edge of the "spiral", ***all*** of these individual spirals are connected to the lead (wire). This supposedly creates charge/discharge paths of uniform and identical length which eliminates phase shifts and time delays that come from the unequal length paths in caps that use a continuous foil.

These are a little more delicate than more typical caps, but not ultra fragile.

They are sold by TRT. TRT has some worthwhile info sheets available — ask for them. One of them shows recommended applications for audio equipment including power supplies and crossovers. One drawing shows a recommended method of battery biasing caps in a crossover — this works so damn well I am at a loss for words to describe the results. I did a modified version of this on my crossover and man did it make a ***huge*** difference — replacing already battery-biased [highly regarded multiple section film and foil capacitors]. I am talking ***not*** subtle — very large and very significant improvement in transparency, high freqs, detail, dynamics, microdynamics, immediacy, etc.

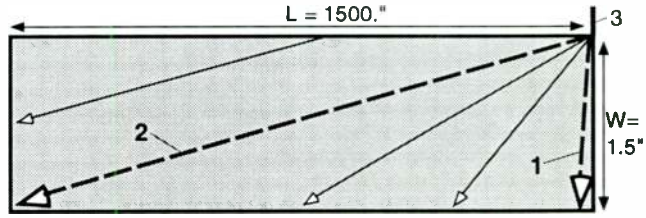
These caps are well worth the price/performance. I look forward to many months of happy experimenting with these caps in other components.

* The original posting explicitly states the capacitor name.

"The Audiophile Network is the granddaddy of cyberspace meeting places for audiophiles, and remains the insider's choice for audiophile information. Dataline phone #818-988-0452." — Stereophile, July 1995

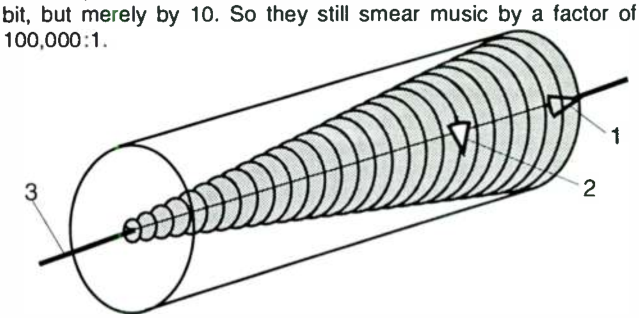
What's Wrong with Your Capacitors?

For any capacitor to work, electric charge must get from the terminal (3) to all parts of the plate. It fans out over your capacitor's plate, following the diagonal paths shown. This creates a bad problem. Path 2 can be 1000 times longer than path 1. So your music doesn't all get through your capacitor at the same time. Some music gets through quickly, via path 1, but some of the same music signal takes 1000 times longer, via path 2. This time smears your music, so it sounds fuzzy, defocussed, and veiled, and perhaps muddy, clogged, honky, or glary.



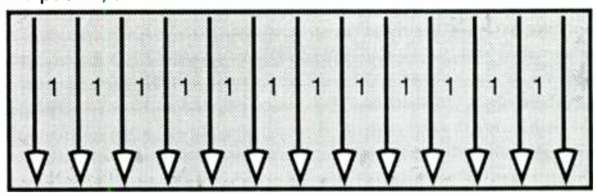
Your capacitors have another problem that's even worse. They roll up signal path 2 into a tight corkscrew coil (below). The inductance of any coil is multiplied by the number of turns squared, and there can be 1000 turns in an audio capacitor. So the inductance of path 2 can be a million times higher than straight path 1, delaying music a million times worse for some paths through the capacitor than for other paths. This capacitor actually smears music by a factor of 1,000,000:1.

Multiple capacitors with 10 sections reduce these problems a bit, but merely by 10. So they still smear music by a factor of 100,000:1.



The New InfiniCap is Different!

Only the new Wonder **InfiniCap®** capacitor cures all these problems. InfiniCap's unique design (patents pend) with metal ends eliminates long, diagonal, corkscrew signal paths like path 2. InfiniCap has an **infinite number of parallel paths**, which are all like path 1, as shown below.



These signal paths are all **short**, so InfiniCap is fast. These paths are all the **same length**, so all of your music signal gets through the capacitor at the same time. These paths don't make coil turns, so they don't have drastically differing inductances.

It's like an infinity of capacitors in parallel – **InfiniCap**! That's why **InfiniCap** sounds:

- transparent, open, and airy — instead of veiled or clogged;
- clean and pure — instead of smeared or dirty;
- clearly focussed, coherent — instead of defocussed or fuzzy;
- fast and delicate — instead of sluggish, hard, or splattered.

InfiniCap gives you all the music at the same time. InfiniCap reveals the subtle inner details, the magic that makes music sound real. Hear the amazing sonic difference yourself.

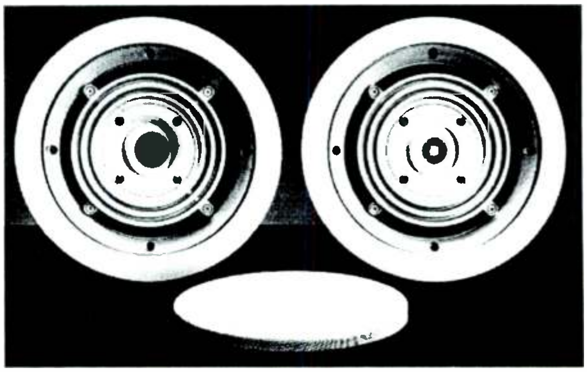
Free sample* InfiniCap!** Write: ***TRT, 408-b Mason, Vista CA 92084 USA

Reader Service #54

Good News

■ TUBING KITS

A comprehensive line of heat-shrink tubing kits is now available. Specially developed "low-shrink" temperature tubing, with special shrink ratios to accommodate a wide range of wires, is standard. The kits can be field-installed using an ordinary hair dryer. Custom printing, tubing types, and colors are also available. Shmarkers, a division of Kimber Kable, Ogden, UT, (800) 746-2753.



■ TWEETER BAFFLES

SpeakerCraft has introduced two 6½" round speakers featuring a new "WavePlane" tweeter baffle that provides better performance by ensuring a smooth transition from woofer to tweeter. The new models, 6.5CRS² and 6.2CRS², are designed for in-wall or in-ceiling installation. They are two-way designs, utilizing a pivoting tweeter for placement flexibility and high-frequency directionality. SpeakerCraft, Inc., 1650 7th St., Riverside, CA 92507, (800) 448-0976.

Reader Service #107

■ DIGITAL SOUND EDITOR

Sound Forge™ XP is a general purpose sound editor for Windows®, with a multiple-window environment that allows more than 50 sound files to be open simultaneously. File support includes Microsoft WAV, Macintosh AIFF, Creative Labs VOC, Dialogic VOX, and Sound Designer SD1. Most Windows-based soundboards are also fully supported. A working demo version of the program can be downloaded from CompuServe (GO SONIC, XPDEM.ZIP) or the Sonic Foundry BBS, (608) 256-6689. Sonic Foundry, Inc., (800) 577-6642.

■ TEST CD

The RRXLO-1000CD is a new XLO test and burn-in CD from Reference Recordings. It includes all the tools needed to set up a system and listening area. The voice of RR's technical director "Prof." Keith O. Johnson is featured. Reference Recordings, Box 77225X, San Francisco, CA 94107, (415) 355-1892, FAX (415) 355-1949.

Reader Service #102

■ PC SOUND SYSTEM

The AD1812 SoundPort® Controller is a complete 16-bit sound system. The chip provides the functionality of five logical multimedia devices: 16-bit sound card, Windows™ sound system, codec, joystick port, MIDI sound, and music synthesis. Sound is synthesized by a DSP core, while Continuous Time Oversampling technology manages sample rate conversion and synchronization. On-chip PC interfaces preserve compatibility with such standards as SoundBlaster. Analog Devices, PO Box 9106, Norwood, MA 02062-9106, (617) 329-4700, FAX (617) 329-1241.

Reader Service #103

■ ACTIVE CROSSOVER UPDATE

Ben Duncan has published a booklet describing how DIYers can update his HFN/RR active-crossover project, the Linkwitz/–24dB/oct PCB. It covers all aspects of upgrading, and new hyper-matched parts are available. Order from Ben Duncan Research, Kirkby Lane, Tattershall, Lincoln, LN4 4PD, England, 44-526-343869, FAX 44-526-342869.

Reader Service #105

■ ULTRADISC II CDs

Mobile Fidelity Sound Lab has announced three Ultradisc II™ releases: Cat Stevens' "Teaser and the Firecat (UDCD 649)," Gerry Mulligan & Paul Desmond Quartet (UDCD 648), and Jean Michel Jarre's "Equinoxe (UDCD 647)." These discs are mastered with a proprietary technology in which all transfers are made from original master tapes, and all the original album text and graphics are reproduced. Call (800) 423-5759.

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

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About This Issue

The subject of this issue's lead article appears on the cover: **Daniel Patten**'s ribbon/planar speaker, which is not only attractive, but is also capable of high-quality sound. Traditionally, constructing your own ribbons or planars was a daunting task, but the author has developed a design that amateur DIY speaker builders can easily produce ("A Push-Pull Planar Speaker Quest," p. 8).

Building your own transmission-line speaker can also be difficult. But, as author **Robert Watson** demonstrates, with the proper guidelines and some assistance from two PC design and measurement programs, you can overcome the problems typically associated with this design ("Design a Three-Way TL with PC AudioLab," p. 22).

In recent years, computers have increasingly benefited speaker builders as a valuable tool for determining crossover design, box dimensions, speaker placement, and so on. But the best is yet to come, according to **Julian Bunn** ("PC Sound Overview," p. 18), who envisions that PC-based audio, with the influx of new products and technologies, will become even more critical for sound reproduction.

In Part 3 of his quest for deep bass from small drivers, **G.R. Koonce** presents his final tests and follow-up work to demonstrate the performance of his double-ended waveguide structure in the low-frequency range ("The Waveguide Path to Deep Bass," p. 30).

Two products in the spotlight include AudioControl's C-101 Equalizer/Analyzer and Electronics Workbench produced by Interactive Image Technologies. Contributing editor **Gary Galo** puts the C-101 through its paces and examines the issues you'll consider before deciding to add it to your music system ("Product Review," p. 48). **Edward Aho** unveils the circuit-simulation power of Workbench for designing, testing, and measuring and applauds its ease of use and low cost ("Software Review," p. 53).

Editorial

PREEMPTIVE IGNORANCE

Prejudice is everywhere, but, of course, most of it lives inside us. We all formulate opinions, emotions, and beliefs ahead of our data. Prejudice is an unambiguous word, since its skeleton meanings are there for anyone who carefully looks to see. The prefix before the "-judice" part means before or prior; the latter is Latin for judgment or judging. Prejudgment would be a near synonym.

Prejudice is clearly one of our pejorative terms. Prejudice—as we teach, and are taught—is bad. We can certainly see the effects of prejudice all around us. The simmering prejudices of blacks and whites alike erupted like an ugly stain on the national consciousness after the recent O.J. Simpson verdict. We are, as I write this, watching as our government, in the person of the Commander-in-Chief, sends troops to the ever-troubled mountains and hills of Yugoslavia to keep what are essentially religious zealot groups from continuing to massacre each other.

Every society, unfortunately, seems to teach prejudicial answers to frightening unknowns. All of us adopt some kind of attitude toward those who are unlike ourselves—those *other* people. We all have unexamined views to unlearn as we try to face our own unexamined response patterns. Our upbringing programs us with automatic responses to not only situations but also to people, as well. Maturity is really a matter of some kind of rational reexamination of how we think we feel about many matters, some of them vitally important to good relationships.

But this is not an essay about other people, it is about ourselves. Our upbringing also subtly conditions us about who we think we are, and what we think we can do. If you are a male, you grow up buttoning your shirt in one way: left over right. Your buttonholes are on the left side of your shirt, your buttons on the right. For a female, the arrangement is the exact opposite. Most young boys go to the barber at some critical point in their early development to have most of their hair cut short. (Yes, I know, this is changing, but the power of the appropriate male hair length is still very pervasive and strong.)

The tomboy and sissy labels for aberrant youthful behavior of girls and boys may not be as common today as they were in my youth, but they are still fairly strong as an undercurrent of expectation. Quarterbacks in the NFL can get away with shoulder-length hair, but officers in the US Marines cannot.

But how has your upbringing influenced who you think you are or what you think you are capable of doing? I suspect that the foundation stones for the attitude that "I'm just not good with my hands..." are buried somewhere out of sight. Somewhere in the early background is a sneer, a too fearful caution, a mockery of some event which went wrong. How recently have you examined your roster of limitations? How did you decide about these inabilities, if at all? When did you decide about things you cannot do?

I am also fairly certain that when people exhibit a lack of interest to the suggestion that they might be able to assemble a computer or learn a new language, we need to reexamine this response. Is it really disinterest? I will certainly agree that some people are genuinely not interested in some matters. But the response may also be a carefully crafted prejudice, based on fear or ignorance.

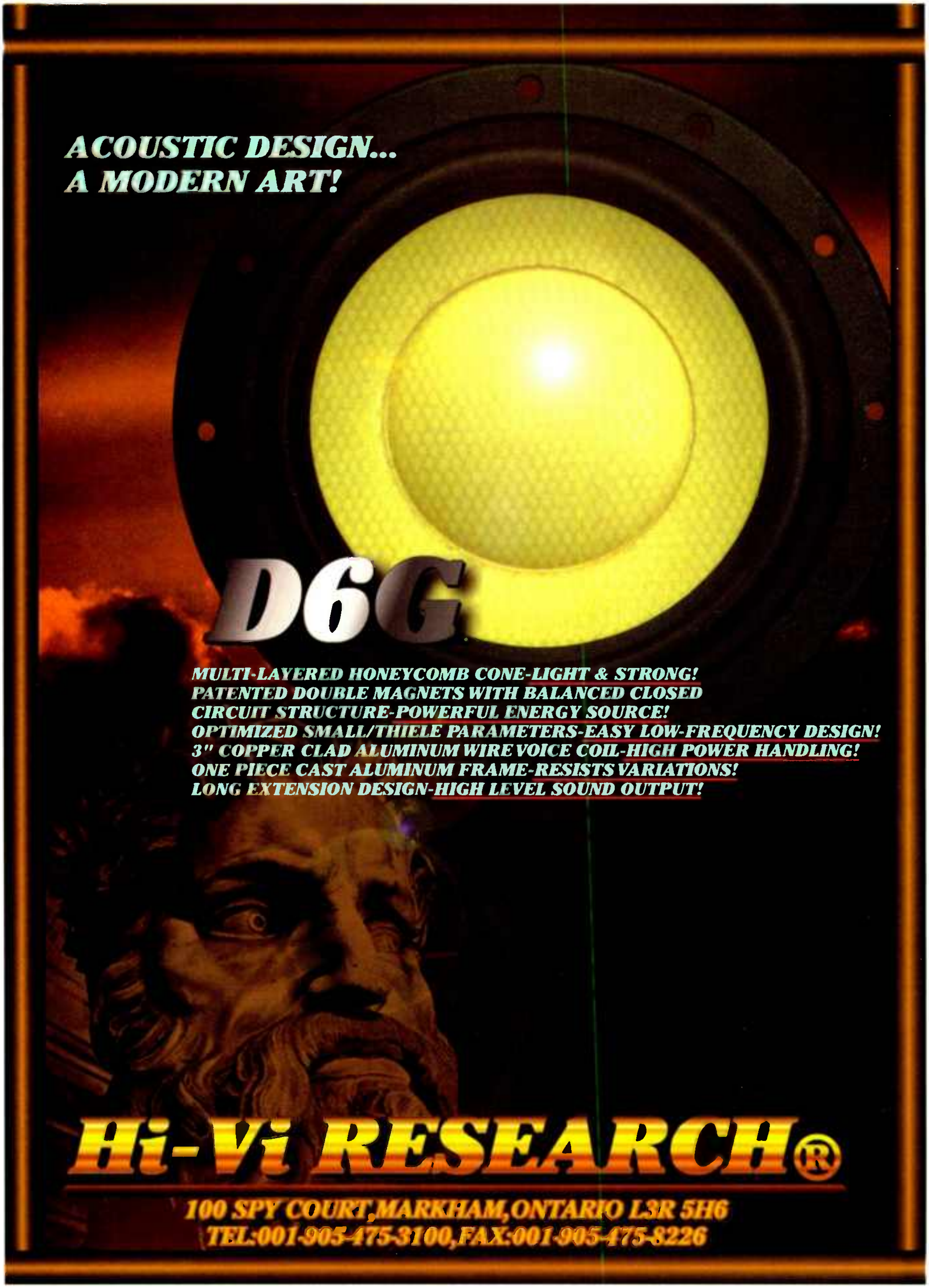
My beliefs about this stem from lots of experiences dealing with a machine or computer program that was not functioning properly. In the course of talking with the person who was trying to use the device, I discovered that in point of fact the object was, for them, a kind of big blur of confusion. Fright can turn off your eyes or your ears, or both. People appear to look but do not see. People appear to hear but are not really listening.

Fear and lack of confidence are psychologically crippling, since they prejudge the relationship to gadgets or programs as being too frightening, complex, or beyond reach. This distorts the true picture of what is in front of the individual. A calmer, more relaxed approach generally miraculously unblurs the picture. The user begins to see individual parts of the machine or the program and understanding dawns.

When this happens to you, it is vitally important to reflect on what has occurred and what it means to your own view of your abilities. One small step of understanding is only the first of many more possible steps. Claim the new vision of yourself, the new discovery of your hitherto unsuspected capabilities.

Again and again, I return to the insights of Martin Buber, who taught us that as long as we regard the universe's particulars as "it" rather than "you" (he used the more familiar "thou"), we are distancing ourselves, perhaps for safety's sake, from the object. Most people seem to think Buber meant this as a comment on interpersonal relationships. A careful study will show he meant an "I-thou" for all of nature. Most of us will never understand a machine which frightens us.

Too many of us have been carefully taught what we cannot do. If we accept those decisions made for us, some with the best of motives, we remain at the mercy of the opinions and prejudices of others. Unless we decide, for ourselves, that such a course and response is one we freely accept and adopt, we are not truly in charge of our own lives. Plato said it best: "The life which is unexamined is not worth living." The helpless, unenabled life is, for me, never free. Have a look from time to time at your own inventory of capabilities as well as inabilities and ask some probing questions. The capability you save will, undoubtedly, be your own.—**E.T.D.**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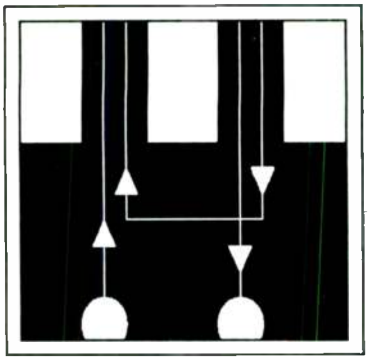
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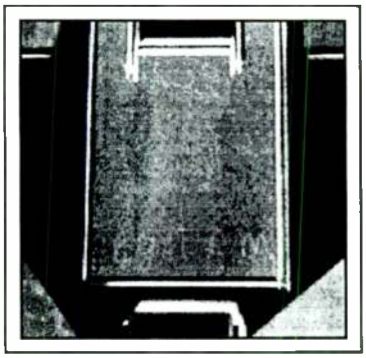
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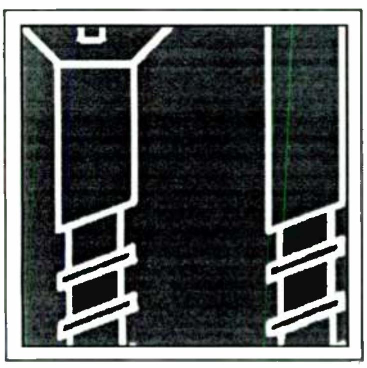
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DEPARTMENTS

Part 1

A PUSH-PULL PLANAR SPEAKER QUEST

By Daniel Patten

Ever since I acquired my first set of ribbon speakers, I have been experimenting with their design. I know many speaker builders who have heard ribbon speakers and really like them. However, it's difficult for the home builder to construct a good set of ribbon speakers. Through much trial and error, I developed some suggestions on how to build a high-quality do-it-yourself WPS40 (Wire Planar Speaker 40″ diaphragm).

BACKGROUND

My first attempts, about eight years ago, included Radio Shack magnets, Plexiglas™, aluminum foil, Handi-Wrap™, and lots of rubber cement. They actually worked for a few minutes! I burned up a lot of plastic in those early years. It was difficult to control my urges to turn up the volume or to ask myself, "How will it sound full-range?" Giving in to these urges always resulted in (poof) my conductors burning through the plastic.

If it wasn't the dreaded burn-up, it was one of a myriad of other problems: all the magnets meeting in the center of my speaker, the plastic stretching, the supporting structure for the magnets bending and falling apart, the amplifier overheating from driving too low of an impedance, or the bond between the ribbon and diaphragm coming apart (this sounds the worst of any of these problems).

Although quite crude, some of my first attempts showed promise as high-frequency speakers. Encouraged by this limited success, I planned to achieve something bigger with my next design: a 1kHz–20kHz response.

AT THE SHOP

I convinced a local sheet-metal shop to help with my next experiment. I drew up rough plans, from which they constructed a unit for about $150 that consisted of 3/32″ sheet steel cut in 3/8″ wide strips 24″ long. They also supplied aluminum cross bars to hold the strips in place on a wood frame. This experiment actually worked, if I handled the speakers very carefully, but if I flexed the metal strips at all, the magnets would pop off. Sometimes, one magnet falling off the structure caused a chain reaction, and several magnets ended up in a pile.

I tried several glues, including Super Glue® and several different types of epoxy. Nothing seemed to work very well.

The first hurdle in constructing a reliable set of WPS40 speakers is to build a strong structure. I talked with a few machine shops and received bids in the $500–$1,000 range! Well, you guessed it. I put the project on hold for a couple of years.

I resurrected the WPS40 project when my employer discovered a company that does laser cutting directly from CAD drawings. I asked a work associate to draw my dream magnet structure in AutoCad (thanks, G.B.) and delivered the plans (*Fig. 1*). The prices were very reasonable.

DESIGN CRITERIA

I designed this WPS40 to solve all my previous problems, and identified the following requirements:

1. Massive structure to keep the magnets a constant distance from each other;

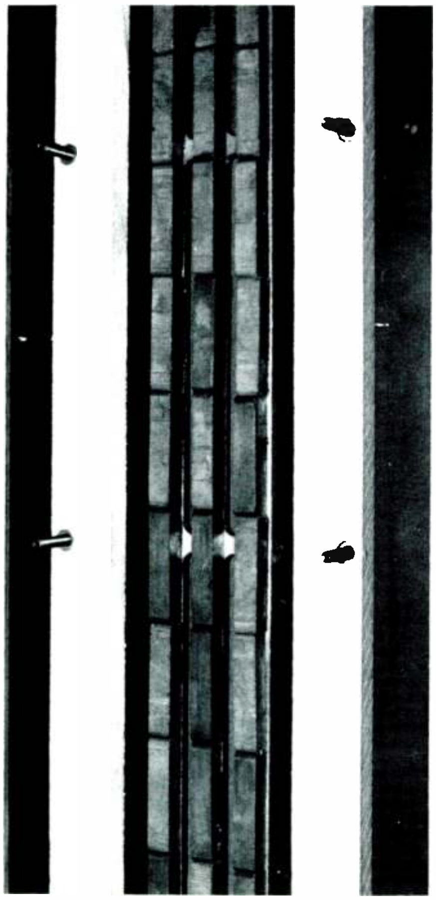

PHOTO 1: Magnets and spacers.

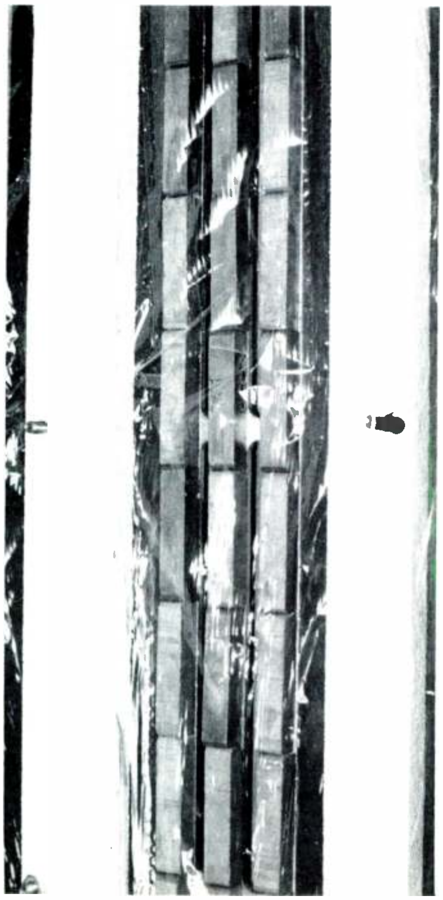

PHOTO 2: Attaching Mylar.

ABOUT THE AUTHOR

Dan Patten currently works at DAS, an electrical engineering firm specializing in very high speed data acquisition. He recently finished the design and prototype of a parallel-processing engine (having 32 individual 32-bit processors) to handle data from a 500MHz ADC. He holds Bachelor of Science in E.E.T. and Master of Technology Management degrees. His hobbies include speaker design and high-end audio electronics design.

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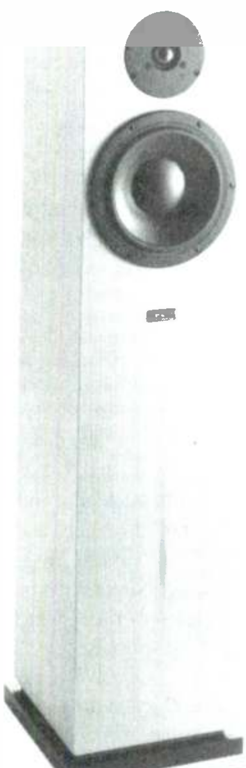


The Aries system showcases the newest Dynaudio loudspeakers: the Esotec 20W75 8 inch woofer, and the Esotec D260 tweeter, in a compact tower design.

The woofer uses a 75 mm voice coil, cast frame, rubber suround, and the tweeter is a chambered, textile 28mm dome unit. In combination with the 6 dB, 13 element filter, this system offers quick response, wide dynamic range, and transients charateristic of live performance.

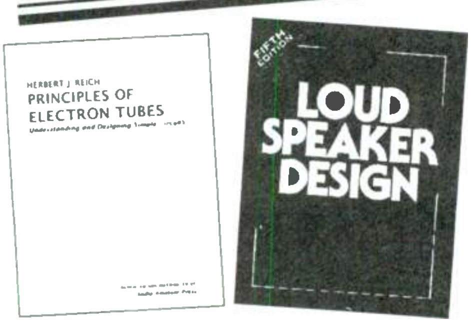
The Aries system is solid, musical, and the prefinished oak cabinets are a pleaure to behold. It has the performance of systems priced at several thousand dollars per pair.

Audio Electronic Supply: SET-II, single-ended class A, solid state, 20 Watt/Channel Stereo Amplifier.

Audio Electronic Supply: SE-1, single-ended 300B Stereo Amplifier.

2 FIRST PRIZES

25 SECOND PRIZES



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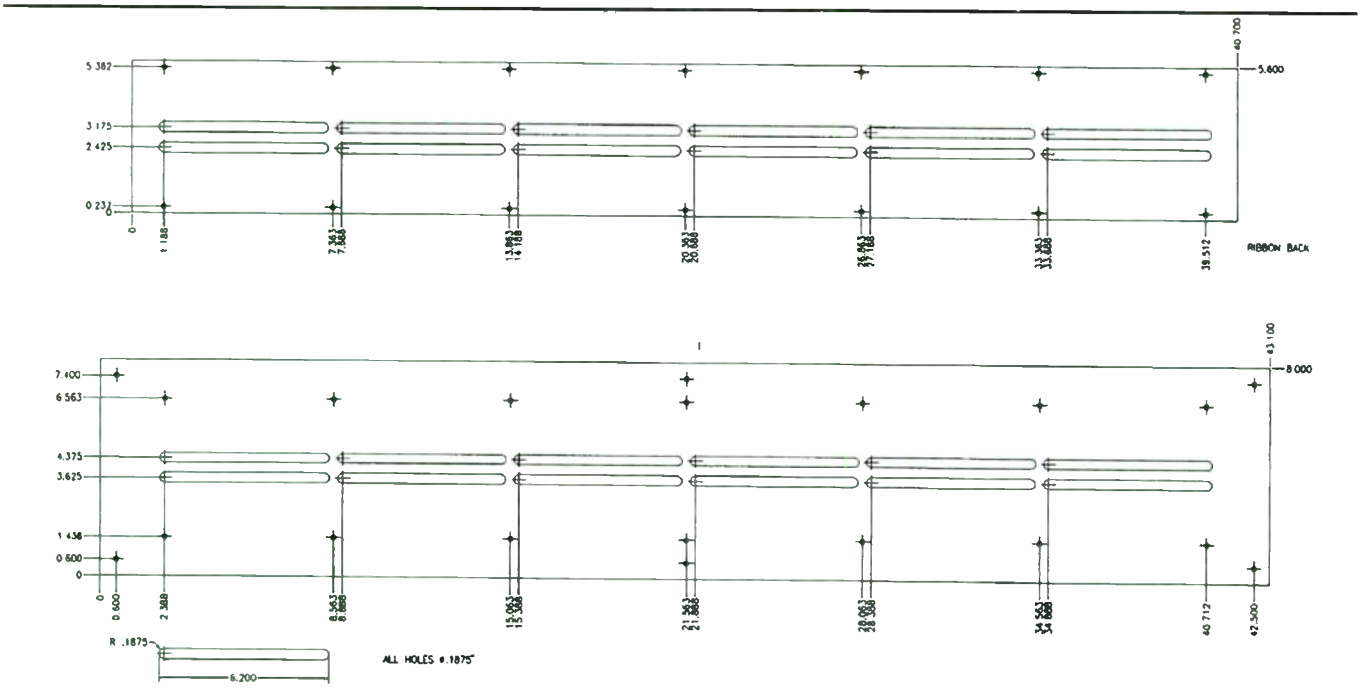


FIGURE 1: Front and rear plates.

2. Rigid, inflexible structure to prevent glued magnets from breaking off;
3. Large diaphragm to reproduce low frequencies;
4. Non-melting plastic;
5. Non-stretching plastic;
6. Conductor bonded to the diaphragm;
7. Increased impedance;
8. Light diaphragm for a good high-frequency response;
9. Speaker efficiency;
10. Powerful magnets at low cost.

STRUCTURE

My solution to the first three problems is shown in *Fig. 1*. I designed this structure for a diaphragm of approximately 40″ and machined it out of 1/8″ steel. I fed the AutoCad drawing into the shop's computer and a very powerful laser sliced it out in a few minutes. The total price was a couple of hundred dollars!

Figure 2 shows a cross section of the plate assembly. Once you bolt this structure together, it is very rigid, not flexing at all. This keeps the magnets in place and prevents them from breaking off. The diaphragm spacers are made with 7/16″ × 1″ wood. I plan to make these spacers out of aluminum for even greater strength.

The WPS40 design uses a 3.5″ wide × 40″ long diaphragm, for approximately 140 in^2 of piston area, which is more surface area than a 12″ woofer (approximately 113 in^2). There is approximately 1/16″ clearance from magnet to diaphragm to allow for plenty of diaphragm movement before it hits the magnet structure. You can use different spacer widths to adjust for either higher efficiency with less clearance or high-power handling capacity with more clearance.

TABLE 1

PLASTIC THICKNESSES

MATERIAL	THICKNESS (1,000s OF AN IN, MIL)
Kitchen plastic (e.g., Handi-Wrap™)	0.4
Mylar®	0.5
Window covering	0.8

TABLE 2

MAGNETIC PROPERTIES

MATERIAL NAME (GENERIC)	MAX. ENERGY PRODUCT
Alnico 5 (sintered)	3.40
Ceramic 5 (barium iron strontium)	3.40
Alnico 5	5.50
Alnico 8	5.30
Alnico 9	9.00
SMCO 18 (samarium cobalt)	16.00
SMCO 20 (samarium cobalt)	20.00
NDFEB 27 (neodymium iron boron)	27.00

PLASTIC

Handi-Wrap™ is inexpensive, and actually worked quite well for my initial experiments because it is very thin and can readily be stretched over the diaphragm spacers. However, it also melts and seems to stretch over time. I also tried the window-covering plastic found in hardware stores. This is convenient to work with because you can use a heat gun to make sure the diaphragm is tight.

My current choice of plastic is Mylar®, which has a very high tensile strength (will not stretch over time), while still being very thin. This is my solution to problems 4, 5, and 8. *Table 1* lists some thicknesses of other plastics I tried.

CONDUCTOP

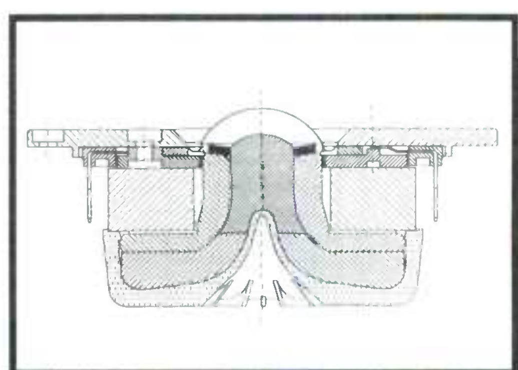
I originallv experimented with foil conductors, using kitchen-type aluminum foil, with an average thickness of 0.8 mil, and metal foil capacitors. These worked well, but gluing the ribbon to the plastic was labor-intensive. Also, the ribbon would detach, and it was difficult to achieve a suitable impedance. I experienced limited success using 1 mil thick and 7/32″ wide copper foil, which I purchased from a stained-glass shop. This foil is handy because it is adhesive-backed. However, it was a little thick, and reduced the high-frequency response because of its high mass.

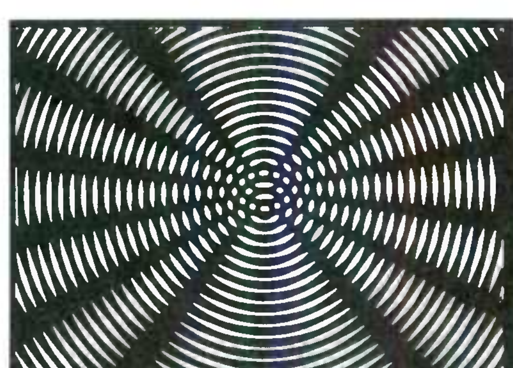
Why use foil? Why not wire? The speakers are not true ribbon speakers because they have a diaphragm. It seemed to me that wire would be easy to work with, and desired impedances were achievable. But how could it be bonded to the diaphragm? I used thin 0.5 mil packing tape, which worked very well (*Fig. 3*).

Using wire and tape solved several problems (6, 7, and 8): it is easy to achieve any desired impedance; you can make as many turns as desired in the magnetic gap; manufacture is simple; attaching the wire to binding posts for the amplifier interface (a difficult task with aluminum foil) is simple; and the wire is very light.

Reader Service #24 →

Music Forever





MAGNETS

Powerful magnets and their proper alignment are paramount to achieving high sensitivity. I arranged the magnets in this structure in a push/pull fashion (*Fig. 2*). Remember the "right-hand rule" when arranging the magnets in relation to current flow of the conductor. Some magnetic properties are listed in *Table 2*.

I used Ceramic 5 magnets with a dimension of 3/8" × 3/8" × 1.875" from The Magnet Source, Model CB-65E. My design requires a total of approximately 120 magnets per speaker, with a cost of $45 per speaker. You can use more powerful magnets; however, they are typically more expensive. If you used the Alnico magnets, the cost would be approximately $400 per speaker, almost ten times the cost for twice the energy product. If you wish to build this speaker with samarium cobalt or neodymium magnets, it would cost thousands of dollars.

To make sure all the magnets are correctly oriented, I used my trusty Boy Scout compass to determine the poles of the magnets, which must be polarized along their lengths with the poles perpendicular to the speaker diaphragm (*Fig. 2*). When arranging the magnetic structure, I had trouble gluing the magnets correctly. I suggest arranging all the magnets for proper orientation *before* starting the gluing process. *Figure 4* shows my procedure to determine the poles of each magnet.

As I mentioned earlier, there are several trade-offs in clearance between the magnets and the diaphragm. I determined that 1/16" clearance was acceptable for adequate sensitivity. (Reducing the clearance seemed to increase the sensitivity.) However, more problems than diaphragm displacement exist. The magnets I received had some tolerance problems; I rejected several because they were too thick (the more clearance you allow, the easier it is to put together).

COST

My prototype speakers' costs were relatively low. The approximate cost breakdown for two speakers is as follows:

Metal work	$300
Wire	1
Mylar	2
Ceramic magnets	100
Hardware (bolts, nuts, banana plugs)	20
Total	$423

At higher quantities, these prices would probably drop by at least 25%.

SOUND

These are some of the better-sounding speakers I have heard, with the open/natural sound that is typically associated with ribbons and electrostatics. I enjoy listening to jazz with saxophones, and these speakers excel in reproducing the sax. In fact, they will probably replace the Acoustat 2200 electrostatic speakers in my listening room. (Anyone interested in a used pair of Acoustats, please let me know.)

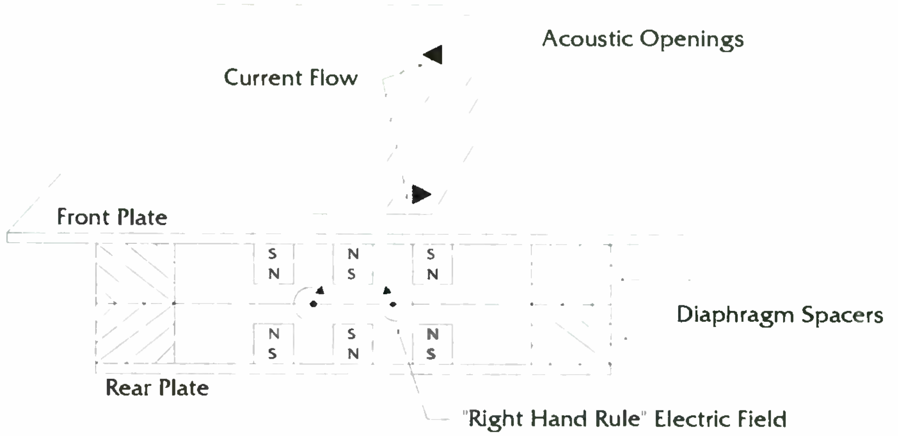


FIGURE 2: Cross section.

The only problem with the WPS40 may be its radiation pattern. As shown in *Fig. 5*, the horizontal dispersion is very good; however, the vertical dispersion is quite limited. This is usually not a problem if you mount the WPS40 so the center of the speaker is at ear level when you are seated.

DESIGN ADVANTAGES

Mass: A typical speaker is pretty heavy and, like a train, once it starts moving it's difficult to stop. Signal overshoot and undershoot are big problems in conventional speakers. The damping factor of the amplifier helps minimize this problem. The mass of the WPS40 speaker is so low it can basically "start and stop on a dime."

Line Source: It exhibits "line source" capability and therefore can add to the realism of the musical presentation.

Push/Pull Design: In both directions of movement, the diaphragm is under full control of the magnetic fields. This typically leads to a more accurate response.

Impedance: Its impedance is purely resistive (i.e., very amplifier friendly). Most amplifiers do not like to drive capacitive or inductive loads. In fact, most amplifiers include a resistor and inductor (yuck) in parallel at their output to compensate for capacitive and inductive loads.

Crossovers: As we all know, coils phase-lead and capacitors phase-lag. So with a cap on the tweeter and coil on the woofer, the signals can be 180° out of phase. This out-of-phase relationship changes as the input frequency changes. If you use a nearly full-range speaker, such as this one, you can keep these phase problems out of the critical midrange and high-frequency areas.

Crossovers: In a conventional speaker, the crossover point between a tweeter and woofer might be 2kHz, *sometimes*. A nominal 8Ω speaker may vary as much as

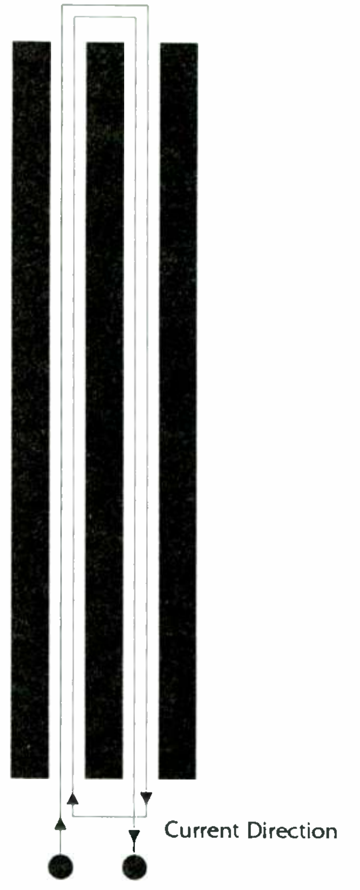


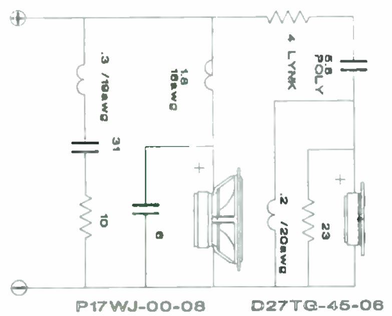
FIGURE 3: Conductor/magnet layout. Keep all current flowing the same direction in each magnetic gap. Two turns with 30 AWG = 1.6Ω, with 32 AWG = 2.2Ω, with 34 AWG = 3.6Ω

LEAP™ Computer Aided Crossover Design

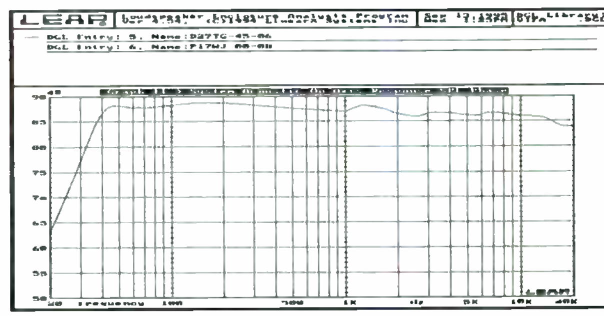
Crossover design has significantly evolved in the last decade. In the past you had to start with a theoretical design and then spend weeks tweaking until you came up with a crossover that sounded pretty good, but probably was not as good as it could be. Madisound has now developed the art of crossover design, providing the customer with a fully developed filter system in a fraction of the time necessary using older methods.

Madisound begins by taking all driver response and impedance measurements in our anechoic chamber using the Audio Precision measurement system. We then export this data into the Leap filter analysis program by Linearx. With the Leap program we can plot what would be an ideal curve for each driver, and then use Leap to try different filter values against the actual driver response curves, continually selecting those parts that bring the actual curve to that of the ideal curve. We then look at the total system response curve and select parts to match the ideal system response. The result is a flat response curve, to the limits of the selected driver responses. This method can be repeated for impedance correction to create an ideal impedance magnitude response. The finished design is excellent, and a good value!

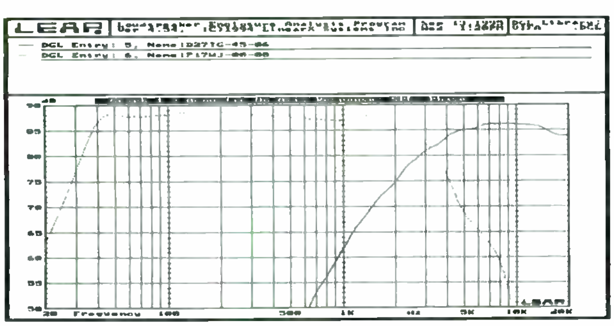
The following curves are provided with each Leap design order:



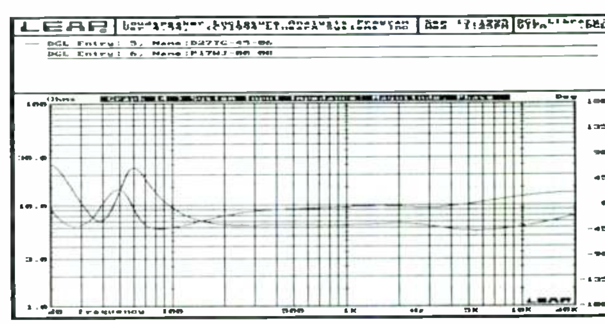
Crossover Schematic Diagram



System Acoustic On Axis Response



Separate Acoustic On Axis Response



System Input Impedance: Magnitude, Phase

To place an order for a Leap design, Madisound we need to know the following:

› What drivers are you using? They must be Madisound stock items. (In the case of 3 or 4 way systems, we could just use the woofers technical parameters.)
› What are your box volumes, or do you wish us to determine them for you?
› Are the drivers surface mounted, or routed into the cabinet for a flush mount? (Flush recommended)
› What slopes (6dB, 12dB, 18dB, 24dB) would you like us to design for, or would you prefer we choose?
› What quality of inductors will you use? We will model using the DCR of the coils.
› Do you want us to design for a flat impedance above box resonance? Additional $5.00 charge.

Leap Crossover Pricing

2-Way Design	$25.00	Subwoofer Design	$35.00	Special requests or projects?
3-Way Design	$40.00	MTM 2-Way	$35.00	Ask about pricing.
4-Way Design	$55.00	Add $5.00 for impedance correction.		

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

Reader Service #1

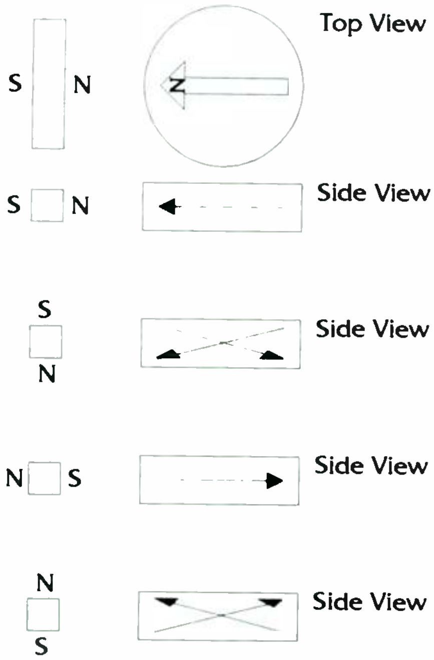


FIGURE 4: Magnet pole orientation.

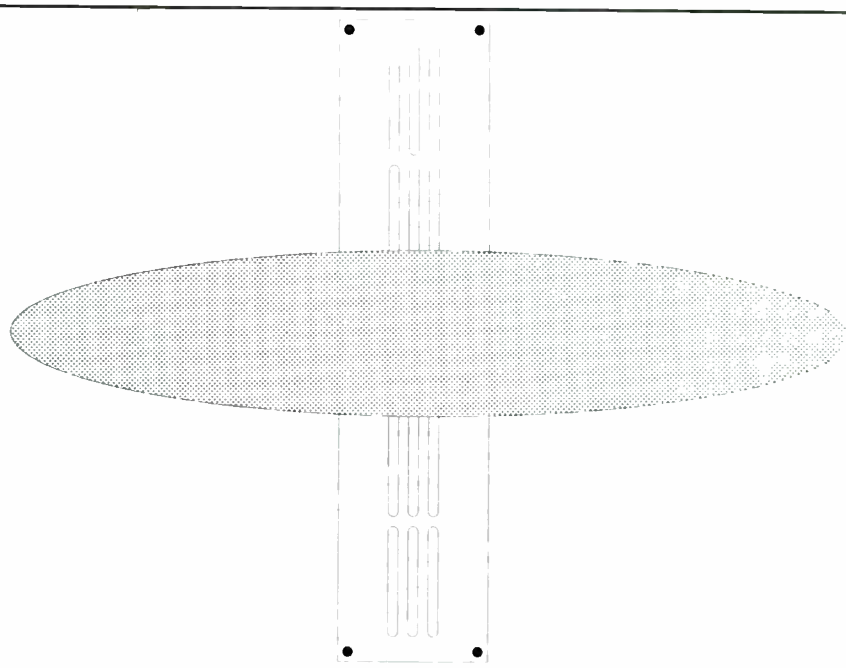


FIGURE 5: Radiation pattern.

2–30Ω. Given the tweeter crossover formula: F = 1/(2 × 3.14 × C × Z), the crossover point will change, which will also affect the speaker's radiation pattern. This planar speaker allows the single crossover point to be below 300Hz.

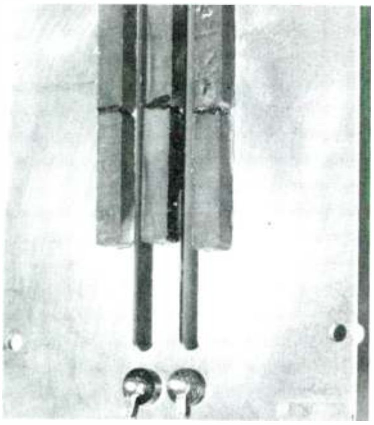
CONSTRUCTION

1. Glue the magnets to the metal plates. As shown in *Fig. 4*, find the poles of all the magnets you are going to use. It's best to line up all the magnets on the sides of the plate and hold a magnet above the rows to ensure the magnetic field is the same for the full row of magnets. Arrange the rows of magnets so when the front and rear plates are assembled, the faces of each row have the same pole (*Fig. 2*). It is paramount to follow the "alternating pattern," which keeps all of the diaphragm moving in the same direction.

Before you glue the magnets to the metal, clean the surfaces with solvent. As shipped from the wholesaler, the magnets are quite dirty with an oily film, which prevents any glue from working very well. I used a paint thinner and a lot of rags to clean the magnets.

2. Attach the spacer to the front plate. I countersunk the nuts in the spacer to accommodate the diaphragm. I used 1.5″ 10–32 hex cap bolts, which are easy to work with, and the exposed cap looks quite nice. The front plate with magnets and spacers are shown in *Photo 1*.

3. Brush the rubber cement onto one spacer, and quickly place the Mylar, which can be punched over the exposed bolts (*Photo 2*). Place the rear-plate spacer over the Mylar and tighten down with nuts. Now do the other side and pull it tight. Wait approximately one hour for the glue to set. Use a heat gun or hair dryer to remove the rest of the wrinkles.



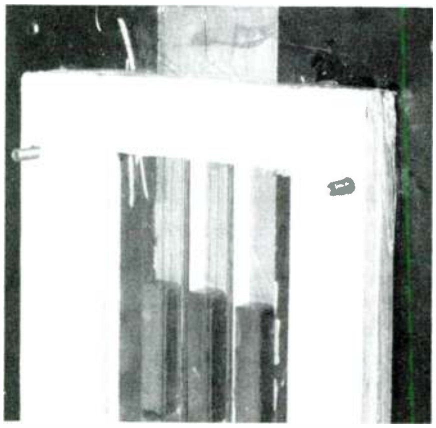


PHOTO 3: Binding post connections.

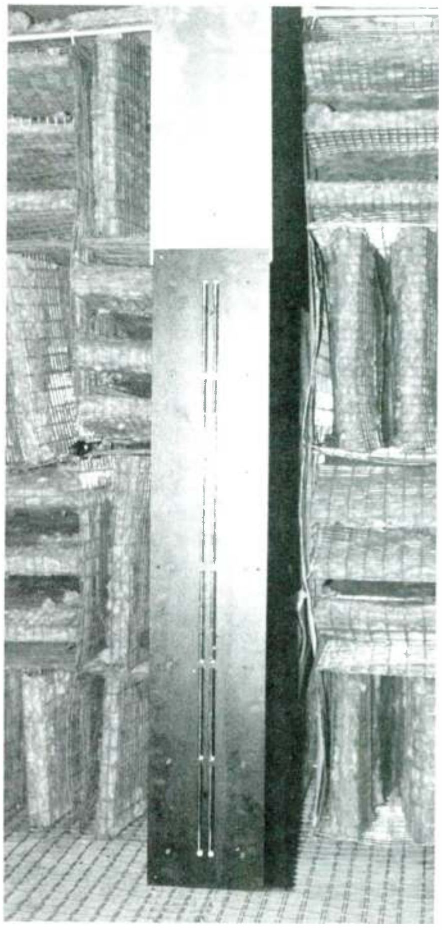


PHOTO 4: Anechoic chamber measurements.

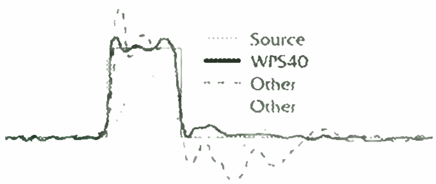


FIGURE 6: Impulse response.

4. Lay down the wire as shown in *Fig. 3*. Thumbtacks pushed into the wood at the ends will hold the wires in place. I used a thin packing tape, approximately 0.5 mil thick and 2″ wide, to laminate the wires to the diaphragm. Then attach the wires to the binding posts (*Photo 3*).

5. The next step, attaching the two plates to each other, is rather difficult, and might require assistance from another person. Since all the magnets are the same pole, the two plates do not go together easily. Start by fastening the end with the binding posts and work your way down to the other end. Tighten all the nuts down at similar rates.

6. Mount the WPS40 speaker in some type of enclosure so the center of the panel is near ear level. Use either a sealed or open enclosure; I prefer the latter. The WPS40 provides a low-end frequency response of approximately 400Hz in open air, and down to 300Hz in my frame. The lower frequency response in the frame is due to the longer path the rear sound-pressure wave must travel before it cancels with the front wave. *Photo 4* shows my finished prototypes in an anechoic chamber. *Photo 5* is the finished WPS40 speaker in the prototype frames.

MEASUREMENTS

Preliminary measurements are:

Frequency response	400Hz–18kHz ±3dB
Impedance	3.6Ω (pure DC)
Sensitivity	87dB @ 1W/m
Power handling	>200W

Figure 6 shows an initial measurement of the WPS40 speaker as compared to a reference square wave and other popular speakers. This measurement reassured me my ears and ego weren't deceiving me.

I have access to an anechoic chamber and should be able to make some interesting measurements and comparisons of planar speakers to conventional cones. I plan to compare the WPS40 against other well-known raw drivers, such as the Dynaudio D54, Dynaudio D52, Dynaudio D28, Dynaudio 21W54, Focal 5K013, Audax HD100 series, and Audax MHD12P25FSM. Part 2 offers these measurements as well as other technical results of this design.

I hope this article helps builders interested in planar-speaker design, and I invite comments or suggestions regarding this speaker, which is capable of high-quality sound reproduction.

SOURCES

The Magnet Source
(800) 525-3536
Magnets (240 pieces CB-65E)

Newark
(801) 261-5660
Wire 30c 34 AWG (part #36F1320)

Radio Shack
Binding posts (two sets)

Available from author:
Mylar plastic*
Front/back metal plates
Wood or aluminum spacers (or from a hardware store)
I will supply, free of charge, the front/back metal plate design in AutoCad or DXF format for the cost of the disk and shipping charges.

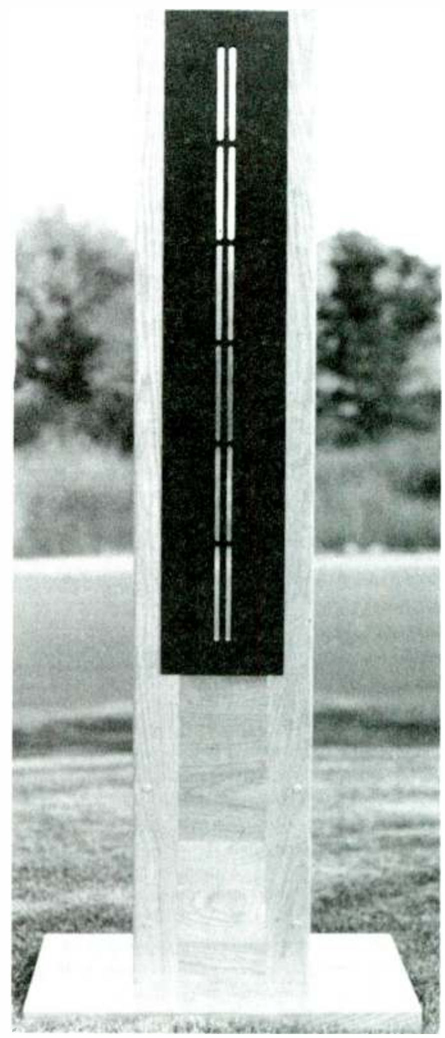


PHOTO 5: Finished WPS40 speaker in frame.

Q: What Makes MIT® MultiCap™ Different from All Other Capacitors?

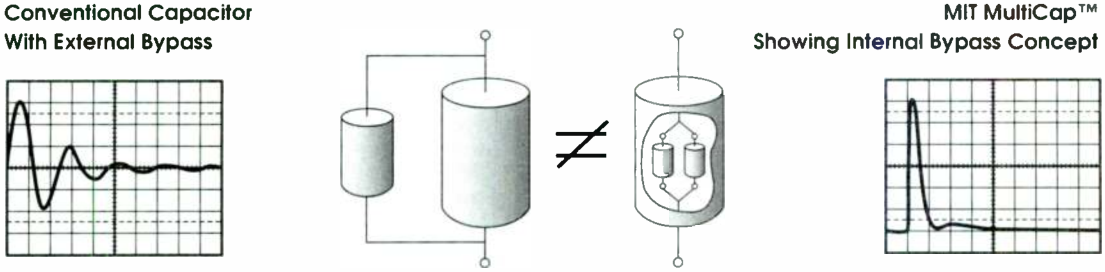
A: Performance. MultiCap's internal bypass design and superior construction deliver higher current capability, lower distortions and noise, greater dynamic range, better phase response, and more control of resonances.

The design of all capacitors other than the MultiCap — including the "special" capacitors that have recently come on the market — conforms to the long-time conventional standard, that is, single-wind metalized or film & foil. The design is called "extended foil construction." Some caps have stranded leads or expensive lead materials. Some have exposed metal ends. None of these alterations represents a significant change in basic design; hence performance will be similar among those caps, varying only with type and quality of materials used in the foil and the dielectric film. And in the integrity of construction techniques.

Metal-end caps, for instance, prove on testing and examination to be the typical single-wind film and foil capacitor — except that they do not have hermetically sealed ends. Degraded performance can be expected as moisture and contaminants in the air enter the dielectric through those non-sealed ends. Try the following test: Dip a metal-end cap and a MultiCap into water and then perform an Insulation Resistance Test. The MultiCap will not be affected, but the unsealed cap will short out — as it can when a user installs it on a PC board and then uses a fluid to clean the board. Copper alloy end material can also oxidize rapidly, and the copper and capacitor plate material are dissimilar metals that will corrode and generate false and unwanted signals due to electrolysis.

MIT uses the same basic extended foil construction, but with many significant enhancements that improve the performance in a way no other capacitor can (the techniques are patented). Note that a characteristic of any conventional capacitor in A/V use is that it **must** be bypassed. The other capacitors in use in audio have to be bypassed, externally, as some of their literature admits. The MultiCap alone is already bypassed internally (up to ten times), and we recommend further bypassing only for the very largest values. The MultiCap is not a single cap of 1000 + turns. It is ten smaller-value **high-speed** capacitors all wound in parallel to make up the total stated value. In other words, a 1 microfarad MultiCap is made up of 10 complete .1 microfarad capacitors joined internally in parallel. This is called **"internal bypassing,"** or sometimes "self-bypassing." One MultiCap is like ten standard caps in parallel. And made with superior construction techniques.

Conventional Capacitor With External Bypass | **MIT MultiCap™ Showing Internal Bypass Concept**



Multiple resonances in a 2 µf capacitor bypassed with an external .22 µf capacitor. Result: high distortion.

Single resonance of the **high-speed,** internally bypassed MIT MultiCap. Result: low distortion.

Externally Bypassing a Conventional Bypass Does Not Equal — in Measured or Sonic Performance — MIT's Patented Internal Bypassing, Which Further Reduces Distortions

The artist's conception (right, above) illustrates two of the MultiCap's up-to-ten internal (bypass) sections as if they were small separate capacitors. The graphs at the sides of the illustration represent measurements performed at the MIT Laboratory on a good-quality film & foil capacitor, bypassed externally, compared to a film & foil MultiCap. MIT MultiCap™ is protected by US Patent # 4,638,402.

MIT ELECTRONIC COMPONENTS

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PC SOUND OVERVIEW

By Julian J. Bunn

Desktop computer multimedia is currently undergoing a technology explosion. This trend is evidenced by the wide variety of hardware add-ons, from sound and video frame-grabber cards and CD-ROM drives to purpose-built desktop loudspeakers. Software offerings include multimedia application authoring tools, voice synthesis and recognition tools, and image-processing packages.

This article provides some background material that I hope will be thought provoking, with some speculation on how the technology advances may affect the role of the PC as a tool for amateur speaker builders. At the end of the article, I predict how these areas of technology will merge and evolve into an integrated home multimedia system based on a sophisticated audio/video computer. Such a system may be used to correct for deficiencies of room acoustics and loudspeaker performance, or for placement in real time.

DIGITAL SOUND PROCESSING

Exciting new products have appeared in the domestic digital audio marketplace over the last few years, in an area once dominated by CD players. Witness the burgeoning field of home cinema amplifiers. Their sophisticated on-board chips allow almost unlimited creativity in terms of digital signal processing and variously artificial sonoric effects.

At the same time, the range of products for producing sound on home computers has grown prodigiously. It is estimated that over half of all home computers currently being sold are capable of generating and recording sound. These computers typically contain extra printed circuit cards designed specifically for this purpose.

A few years ago the electronics and software techniques used to treat sound were of low quality: typically, 8-bit mono samples, with sampling rates of 8kHz, and poor signal-to-noise ratio. This situation has improved considerably. Today the norm is 16 bits per stereo channel, rates up to 44kHz, and signal-to-noise ratios and response curves that qualify such cards for inclusion under the hi-fi "umbrella."

The techniques for generating and recording sound have also advanced. Home computers no longer sound like an old 78 played across a bad phone line. Methods for compressing audio and video data are well advanced, too. Coupled with cheaper, faster, and larger capacity hard disk drives, satisfactory sizes of audio/video tracks or clips can be stored within the computer.

Finally, the speed of CPU chips, the greater memory bandwidth afforded by wider (more bits) and faster buses, and the larger amounts of RAM that are usually installed, all result in sufficient data rates for jerk-free multimedia. Interesting possibilities abound in the area of processing the audio and video digital signals in real time.

THE SOUNDBLASTER STANDARD

The generic PC is a desktop device running MS-DOS, with maybe Windows 3.1 or '95, with an Intel (or clone) x86 architecture chip as the CPU. PCs such as this far outnumber any other type of computer, whether for home or office.

Of course, computer sound is not the exclusive province of the PC; it is standard on the Macintosh, as well as the Sun, and most other high-end "workstations." (Silicon Graphics desktop workstations even include a small video camera, as well as a microphone and loudspeakers, for use in video conferencing.) This article concentrates on the PC specifics of sound-card technology.

The *de facto* audio standard for the PC is SoundBlaster. Manufactured by Creative Labs, a Singapore-based company, this card was first introduced in the 1980s. Being "SoundBlaster-compatible" is a major marketing consideration for other card manufacturers, since vast numbers of PC games require it.

The SoundBlaster standard includes the specification of a certain set of programmable registers, which perform such functions as receiving command strings from the application, returning information on the card setup to the application, setting the play/record mode, altering the mixer settings, and starting and stopping DMA transfers. Creative Lab's stranglehold is unlikely to last indefinitely, as newer operating systems allow programmers to shield themselves from the hardware details through appropriate software "drivers."

DIGITAL SOUND BASICS

Since PCs cannot directly manipulate analog signals, they must deal with digital units that are generally multiples of an 8-bit byte. Both Analog-to-Digital (ADC) and Digital-to-Analog (DAC) converter chips must be present to convert the signals to and from a usable format.

The basic sound-*generation* operation converts the value represented by one byte into a voltage level. Since an 8-bit byte can represent up to 256 (2 to the power of 8) different values, the voltage level generated can have the same number of values.

Conversely, the basic sound-*recording* operation converts a voltage level into a byte value. By stringing together a series of bytes, each representing a different voltage level, a sound waveform is emulated. Manipulating the string of bytes in various ways alters the resulting sound wave.

The term Digital Signal Processing (DSP) refers to the methods by which signals are treated algorithmically. Sound synthesis is a special DSP technique for generating a digital signal which, when converted to analog and played through a transducer, sounds like a musical instrument. This technique is especially important in the development of sound-card technology, and various methods are commonly used.

FM VS WAVE TABLE SYNTHESIS

FM synthesis is based on the concept of "operators"; the more operators, the more satisfactory the synthesis. One or more sine tones ("carriers") are modified with one or more operators. The frequencies of the operators determine how the carriers are modified: the resulting sound is frequency modulated.

This very general technique allows not only the emulation of traditional sounds, but also the generation of completely new sounds. One disadvantage is that the synthesized sounds of real instruments are rarely very realistic. The initial SoundBlaster cards sported FM chips which were made (at that time) only by Yamaha. Since FM synthesis

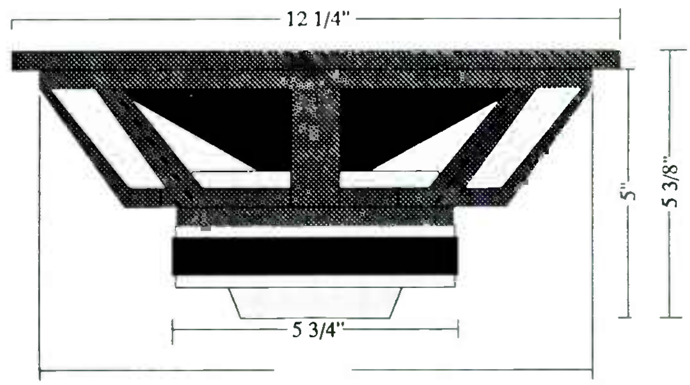
NHT 1259

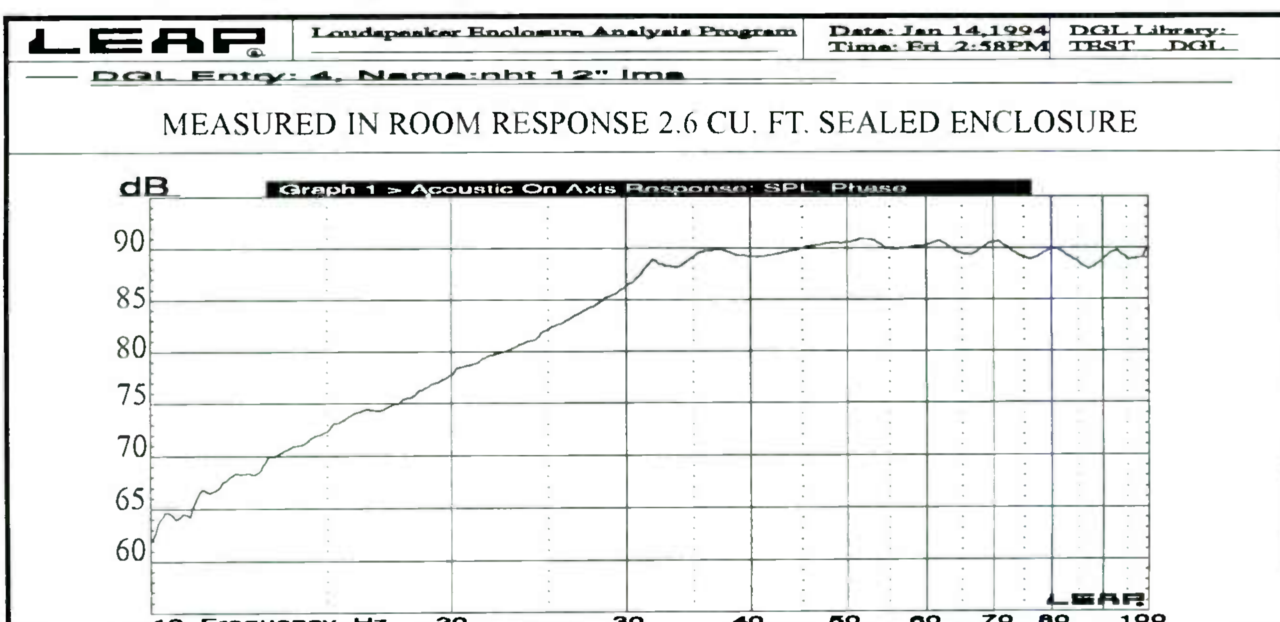
NOMINAL IMPEDANCE	4 ohm
MUSIC POWER	300 W
FREQUENCY RANGE	19-1000 Hz
SENSITIVITY (2.83 V)	90 dB
EFFECTIVE CONE AREA	507 cm2
VOICE COIL RESISTANCE	3.4 ohm
VOICE COIL INDUCTANCE	1.2 mH
VOICE COIL DIAMETER	50 mm
VOICE COIL HEIGHT	34 mm
AIR GAP HEIGHT	8 mm
XMAX	13 mm
FREE AIR RESONANCE	19 Hz
MOVING MASS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

A&S Speakers is pleased to be able to offer this custom made, high performance woofer from Now Hear This (NHT). This is the cost no object woofer used in the highly acclaimed NHT 3.3 system. Unlike almost all other woofers, the NHT1259, with its low resonance and exceptionally long voice coil is designed to produce deep bass in a small sealed enclosure when positioned within two feet of the rear wall. This design results in superior transients and minimal phase shift and group delay. Every construction feature contributes to the outstanding performance of this driver: the 50mm diameter, 34mm long, 2 layer voice coil allows long excursion, high thermal capabilities and tremendous output; the large bumped backplate and raised spider prevent bottoming at maximum excursions; the heavy cast frame minimizes energy transfer to the enclosure; and the rubber surround and polypropylene cone promise long term durability in every environment.

Price: $149.00/each

***Cabinets and plans for our computer designed enclosure utilizing 14 internal braces are now available upon request.**





A&S speakers

4075 Sprig Drive
Concord, CA 94520
(510) 685-5252 Fax (510) 603-2724

is required for SoundBlaster compatibility, it is likely to be around for some time.

Wave Table synthesis, in contrast, allows extremely faithful simulation of real instrument sounds through digitized recordings in the form of "wave files." Boards offering Wave Table synthesis usually include a selection of available instrument sounds. If a new sound is needed, it can be downloaded from a file repository accessed over a network via modem, or purchased on a diskette. A disadvantage is that it requires a lot of RAM to store the Wave Tables. Some chips, such as the Yamaha OPL4, have permanent ROM that contains those commonly required.

DIGITAL SIGNAL PROCESSING

The inclusion of one or more DSP chips (as opposed to FM or Wave Table chips) on the sound card will almost certainly be the future means of handling computer-based sound. The key advantage of this technique is that it gives the developer full control over how sounds are generated or treated. It does not rely on a fixed method based on chip logic, as in the case of FM or Wave Table synthesis.

DSP chips are programmed to apply an algorithmic process to a digitized audio signal, or to generate an audio signal directly.

The details are then left up to the application designer. Examples include addition of reverberation to an existing signal, application of an FFT for voice recognition purposes, and sound simulation of each digit on a touch-tone telephone.

Two disadvantages with DSP are that the chips tend to be expensive, and they are not easily programmed. Both of these objections, however, are likely to become increasingly less valid.

SOUND CHIPS

The heart of the sound card, then, is the chip set that controls the DSP. Audio DSP reflects a general need throughout the industry for chips that can process digitized signal data at high rates and in great precision. Consequently, the silicon industry is ramping up production capacity, and pouring research and development money into new chip designs. The following examples are specifically tagged for the audio market.

Yamaha's OPL4 combines 20 FM and 24 Wave Table synthesized voices on the same chip. An optional effect processor provides surround sound, echo, and reverberation. The Wave Table data, which is stored in ROM, can be in 8-, 12-, or 16-bit sample format.

The Analog Devices AD1845 chip incorporates full-duplex record and play, and variable-frequency sampling rates. This company's chips are already widely distributed on boards from such manufacturers as Orchid Technology, Hewlett-Packard, and Kurzweil Music Systems. The latter markets a "MASS Sound Engine" with 32-voice Wave Table synthesis, and echo, reverberation, flanging, and pitch-shifting effects.

Texas Instruments' TMS320C30 and TMS320C40 chips are used by a number of board manufacturers who offer PC hardware and software products for analyzing audio signals. Sonitech and Loughborough Sound Images have product ranges which include sophisticated spectrum analysis, speech recognition, and filtering tools based on these chips.

Hitachi is developing chips dedicated to audio/video applications. Intel has announced its intention to incorporate DSP in the next generation of Pentium-based PCs. TriMedia/Philips is working on a multimedia chip known as a programmable DSP/CPU. As its name suggests, it combines CPU and DSP partitions on the same chip, and will probably be programmable in C.

THE VERSATILE PC

Considering the technology trends and equipment already available, the tasks of the domestic PC are likely to diversify in the years ahead. It already comes into its own as a speaker building tool when crossovers need to be designed, box dimensions calculated, optimum loudspeaker positions estimated, and so on. Many such tools exist commercially, in the public domain, or as shareware; however, they are all passive in the sense that you must type in parameters and measurements before the PC can make the required calculations.

More exciting possibilities are now emerging, in which the PC itself gathers the data, then analyzes and displays it in a meaningful form. The obvious examples are several products which turn the PC into an audio frequency-response measurement tool: you just position the microphone and the PC plots the response of the equipment.

Imagine a tool that completely automates the process of designing loudspeaker crossovers. You have built the box and installed the drivers, and you want to build a crossover that produces the flattest response curve (even though this may not produce the best sound). Leave your copy of *The Loudspeaker Design Cookbook* in the bookcase and turn on your PC instead.

This PC contains two sound cards: one each to drive the tweeter and the woofer. Connect the tweeter to the output of one card, and the woofer to the output of the other. Place a microphone, which is connected to the input of one sound card, a short distance from the loudspeaker. Now fire up the automatic crossover-design software, and watch the PC screen display the progress of its deliberations!

After a few moments, it prints out the circuit diagram and LCR values of the optimum crossover for your box and drivers. It asks if you'd like to listen to some music to evaluate what the system would sound like, or alter the order of the crossover to determine whether that would produce any improvement. To do all this, the crossover design software uses a simulation of the response curve of an N-order crossover with given LCR values to send the appropriate (different) signals to the tweeter and woofer. It then records and analyzes the resulting sound.

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Phil Atherton, "Could low cost DSP signal the end for analogue audio?", *Electronics World + Wireless World* (May 1993).

Dennis Cronin, "Examining Audio DSP Algorithms," *Dr. Dobbs Journal* (July 1994).

Loughborough Sound Images, "A Synthetic Concert Hall in the Home," *DSP Link*, Issue 11.

According to this analysis, the software simulates an adjustment of the LCR values to fit a flat response curve. It is a very simple computational task, using a least-squares method, to fit the curves.

A PC IN YOUR HI-FI

Modern consumer audio systems allow you to superimpose the acoustic response of one venue on top of music recorded in a different venue. These systems offer varying amounts of control over the parameters of such audio signal treatment.

The conversion component of this system can conceivably be replaced by an audio DSP-capable PC. The only limitations to the variety of effects that can be achieved are the flexibility of the DSP computer programs running in the PC and your imagination.

A set of software DSP "tools," for example, can be conjoined as building blocks to achieve the desired processing effect or monitoring. Some of the more obvious include:

- an FFT tool that displays the frequency components of the source signal (i.e., a spectrum analyzer);
- a graphics equalizer to boost or suppress bands of frequency in the signal;
- a stereo L-R difference signal extractor used to detect mono, or as an extra output;
- a Dolby Pro-Logic emulator that in software extracts the surround sound information;
- a digital filter, with coefficients calculated to reproduce the acoustic environment of an arbitrary venue;
- a convolver, which convolves the digital filter with an audio signal to reproduce the sound in the venue;
- a room response calculator, used with a microphone, which folds out the source signal from the measured signal and yields the coefficients of the digital filter;
- an inverter, which flattens out the measured room response by using its inverse.

THE FUTURE

One main trend identifiable in PC-based audio today is clearly that of integrating audio DSP on the PC motherboard. This eliminates sound-card installation hassles, and allows the manufacturer to properly integrate and optimize the power of the sound chip set with the rest of the electronics. We are likely to see full-duplex play and record, increasingly sophisticated on-board DSP algorithms (perhaps including multichannel surround sound decoders), higher sampling rates and better filters, and 32-bit sampling, rising to 64 bits in the longer term.

This discussion would not be complete without mentioning devices which will most likely be combined into a single home control center. They include the integration of the telephone system, with the full functionality of an answering machine and a FAX machine, together with voice recognition and synthesis software that can identify the caller and formulate an appropriate reply. Other software allows the caller to interact with the PC by using, for example, the buttons on a touch-tone telephone.

As an integral part of the home telephone system, the PC can provide access to remote computers via dial-up lines, allow you to download or upload audio/video files, and even play them in real time (once compression algorithms have advanced sufficiently).

Access to the Internet, and hence to a vast wealth of on-line information, is another highly attractive possibility. Users of the World Wide Web, with its fully interactive choice of many terabytes of audio/visual information stored in servers across the globe, already appreciate the exciting possibilities of the fully networked home computer. In conclusion, the amateur loudspeaker builder only stands to gain from embracing the PC as another indispensable tool of the hobby, alongside the table saw and the well-thumbed copy of *The Loudspeaker Design Cookbook*.

Part 1

DESIGN A THREE-WAY TL WITH PC AUDIOLAB

By Robert D. Watson

There continues to be an avid group of speaker builders who prefer to fashion transmission-line speakers. Unfortunately for the novice in this area, the results of hard work can sometimes be disappointing, primarily because there are few theoretical models to predict the design's frequency response, the actual construction is very difficult, and tuning the transmission line by stuffing it with sound-absorbent material could previously be done only by ear. This article discusses how I used PC AudioLab and Bullock's TLBOXMODEL to design and measure a three-way, triamplified transmission-line speaker.

INITIAL DESIGN

After reading *The Loudspeaker Design Cookbook* several times, I sketched a design for the speaker. Fortunately, I have a talented brother who was able to transform my rough sketch into reality. *Figure 1* shows the working diagram. As you can see, I used a folded-line design that places the woofer near the floor to minimize the boundary-induced dip in the low-frequency response caused by floor reflections.

I designed the transmission line length to be 2 meters. The initial cross-sectional area behind the woofer is 95 in^2, and the cross-sectional area at the port is 38 in^2, which gives a taper ratio of 2.5. I chose to use a Dynaudio 17W-75 as the woofer because of its excellent power-handling capability, low distortion, and extended-frequency range.

The mid- and upper-frequency ranges are handled by ScanSpeak 8636 cone midranges and a Dynaudio Esotec D-260 dome tweeter, respectively, and are positioned in an M-T-M arrangement. The sides of the cabinet are covered with "Black Hole" damping material to reduce vibration, and eggcrate foam and Acousta-Stuf® serve as the acoustic-line absorbent material.

If money and time are of no concern, a tri-amplified speaker will have lower distortion and higher power-handling capacity than traditional amplifier-speaker arrangements. In my design, the three 120W MOSFET power-amplifier boards are mounted, along with output transistor heatsinks, in a recessed cavity at the back of the speaker. The power-supply transformer and board are located in the base. I used a toroidal power transformer to minimize any electromagnetic interference with the woofer voice coil and magnet structure. Also, the power-supply board contains a 12V power relay to facilitate remote turn-on of the speaker.

THE ALL-IMPORTANT ACCESS

It's important in designing a TL loudspeaker to make access to the TL cavity relatively easy so you can change the placement and

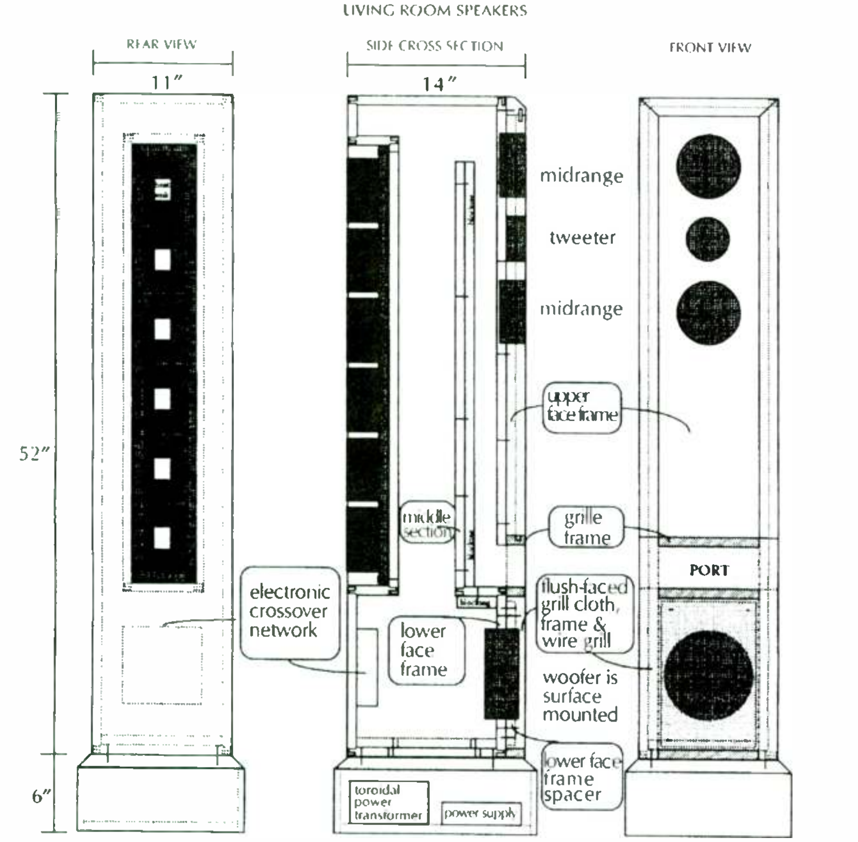

FIGURE 1: Drawing of transmission-line enclosure.

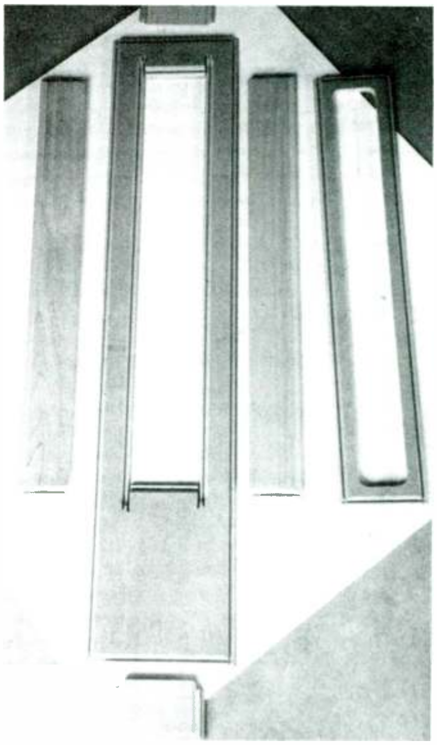

PHOTO 1: Rear-panel pieces ready for assembly.

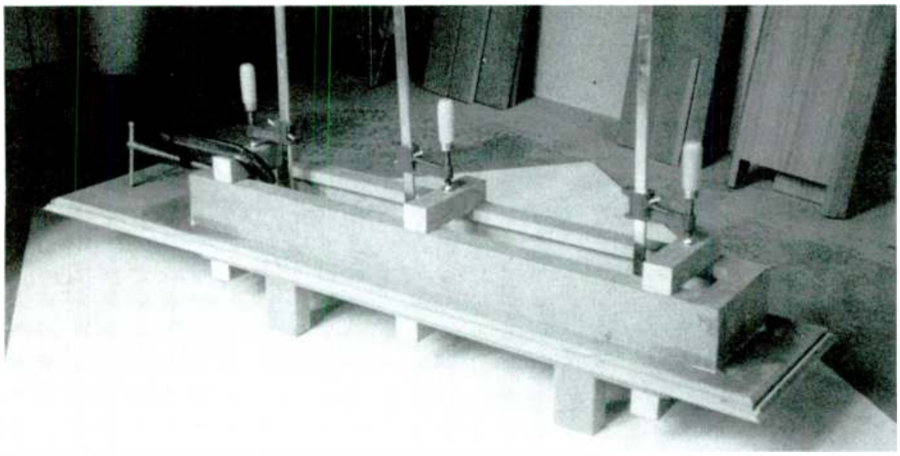

PHOTO 2: Assembled rear panel.

amount of acoustic stuffing. Referring again to *Fig. 1*, the heatsinks are mounted on a removable panel so you can readily stuff the rear TL cavity. The base is also detachable for easy servicing access to the power supply and transformer, and you can stuff the front TL cavity via the speaker-hole cutouts and the port opening. A removable grille covers the woofer and port.

TABLE 1

LOUDSPEAKER DRIVER CHARACTERISTICS

	f_S	Q_{TS}	V_{AS}
published data	39Hz	0.74	18.8
measured data	47Hz	0.69	17.5

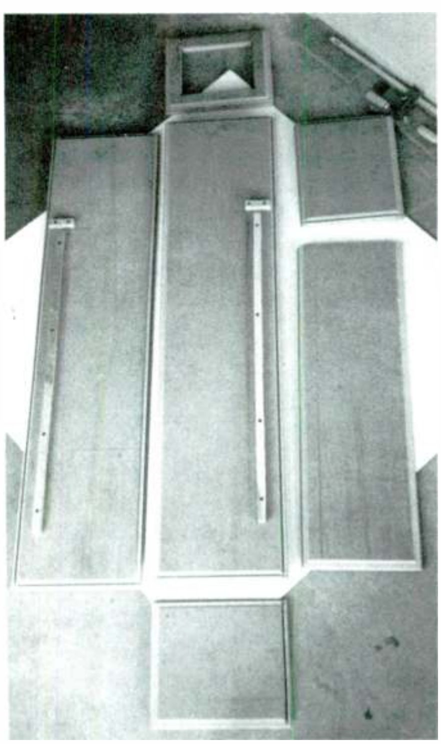

PHOTO 3: Main TL cabinet pieces ready for assembly.

As you can see from *Fig. 1*, construction of this cabinet is not for the woodworking novice or faint of heart. Several construction points may help any brave souls who wish to reproduce this design. We used ¾″ high-density fiberboard with a cherry veneer throughout and a special router bit to create an interlocking miter joint. We first constructed the recessed cavity that holds the detachable panel for the power-amplifier boards and heatsinks, and then attached it to the back panel to create the rear of the enclosure.

Photo 1 shows the unassembled rear-panel parts, and *Photo 2* the completed rear panel. *Photo 3* shows the component pieces for the main TL cabinet. Notice the blocking strips attached to the side walls to aid in attaching the middle section. The top, bottom, front, and side panels are glued together (*Photo 4*) to form the main transmission-line enclosure. Note especially that you do not attach the middle-section panel, which divides the rear and front transmission-line path, until later so you have access to attach the upper-face frame.

Photo 5 shows the middle section in place for illustrative purposes only. *Photo 6* shows the construction of the upper-face frame, which you attach to the front of the enclosure with glue and screws from the rear so that the exposed veneered face does not show any of the construction details. *Photo 7* shows a rear view of the cabinet at this stage of construction.

After attaching the upper-face frame, you can place the middle-section divider using the blocking strips. Finally, you attach the rear panel to complete the job. The completed TL enclosure, without the base, is shown in *Photo 8*.

THEORETICAL RESULTS

TLBOXMODEL is a software program by Robert Bullock, III, that models the low-frequency response of a transmission-line speaker. It allows you to specify the TL

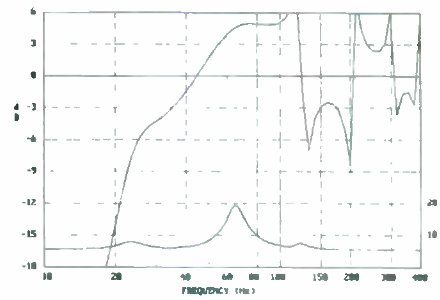

FIGURE 2: TLBOXMODEL impedance and SPL result for 0.1 lb/ft^3 stuffing.

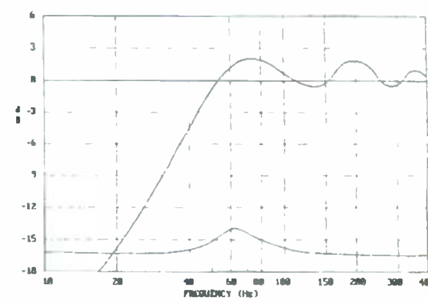

FIGURE 3: TLBOXMODEL impedance and SPL result for 0.5 lb/ft^3 stuffing.

length and taper, the amount of acoustical stuffing, the adjusted speed of sound, and the loudspeaker driver characteristics, such as f_S, Q_{TS}, V_{AS}, etc. *Figures 2–4* show the theoretical SPL response and impedance as you alter the amount of TL stuffing.

Figure 2 shows the response with 0.1

PHOTO 4: Assembled main TL cabinet without rear panel attached.

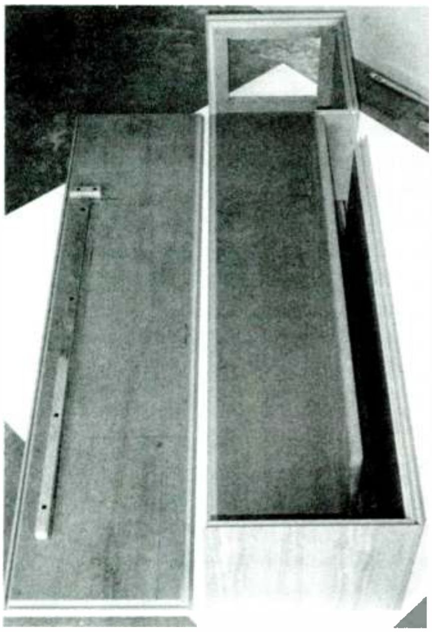

PHOTO 5: Cut-away view of middle-section divider in place.

lb/ft^3 stuffing. As you can see, the SPL response has several resonance peaks and valleys, and the impedance curve also demonstrates several resonance peaks. *Figure 3* shows the effects of adding 0.5 lb/ft^3 of stuffing. The line resonances are somewhat damped, though there is still quite an irregular frequency response. *Figure 4*

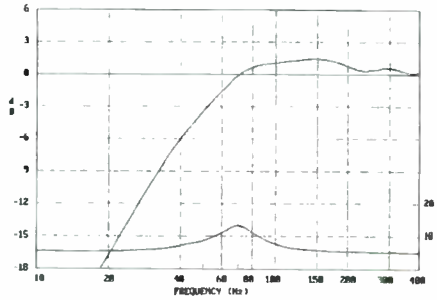

FIGURE 4: TLBOXMODEL impedance and SPL result for 1.0 lb/ft^3 stuffing.

shows the SPL response with 1.0 lb/ft^3 of stuffing. Here, the resonance peaks and valleys have been minimized, but at the expense of some low-frequency response.

Still lacking are accurate values for the adjusted speed of sound in the transmission-line cavity and the loudspeaker driver characteristics. To measure these quantities and the final transmission-line SPL frequency response, I used several measurement instruments available in PC AudioLab.

USING PC AUDIOLAB

This software measurement program uses a computer soundboard to measure audio electronics and acoustics. PC AudioLab supports the Creative Labs SoundBlaster ASP16 and AWE32 soundboards, which have an inde-

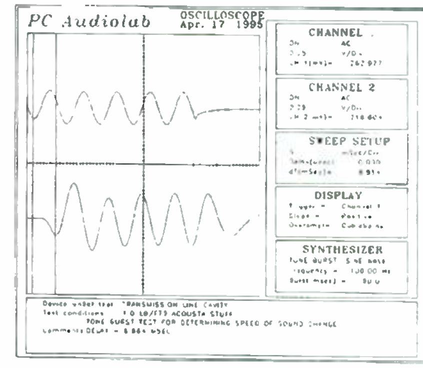

FIGURE 5: PC AudioLab oscilloscope measurement of the speed of sound in stuffed transmission line.

pendent FM synthesizer and stereo 16-bit analog-to-digital converters (ADC), with a maximum stereo sampling rate of 45kHz. The SoundBlaster AWE32 soundboard also has an EMU 8000 DSP for wavetable synthesis, which PC AudioLab uses for white- and pink-noise generation and 32k point MLS analysis.

With this arrangement, the program is able to emulate several electronic and audio test instruments that would cost thousands of dollars if bought separately. PC AudioLab can emulate a dual-channel oscilloscope, a dual-channel FFT spectrum analyzer, an electronic network analyzer, a dual-channel THD analyzer, a loudspeaker analyzer, an

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Speaker Builder regularly receives audio-related literature—catalogs, newsletters, books, and other publications—covering products and topics of interest to *SB* readers. The following is a compilation of recent references and titles:

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2.2	14 / 31	$1.90	**10**	23 / 46	$4.30
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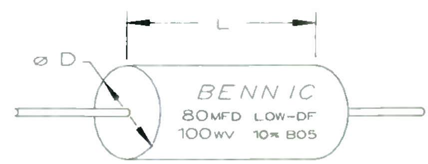
Max D.F.	:	Maximum 5% @1KHz Values 12mfd to 140mfd Maximum 10% @1KHz Values 180mfd to 280mfd
Color	:	Blue



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Value mfd	D / L mm	Price Each	Value mfd	D / L mm	Price Each
12	12 / 31	$0.80	**100**	18 / 42	$2.70
22	12 / 31	$1.05	**125**	22 / 45	$3.10
31	16 / 34	$1.35	**140**	22 / 45	$3.70
40	16 / 34	$1.65	**180**	22 / 45	$3.80
50	16 / 34	$2.00	**200**	16 / 42	$4.00
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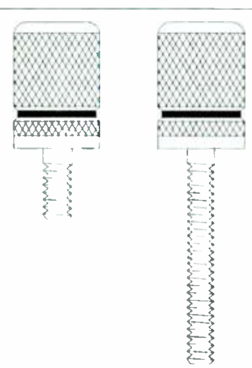

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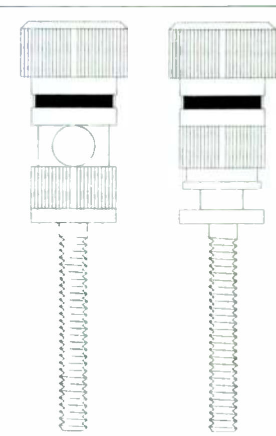

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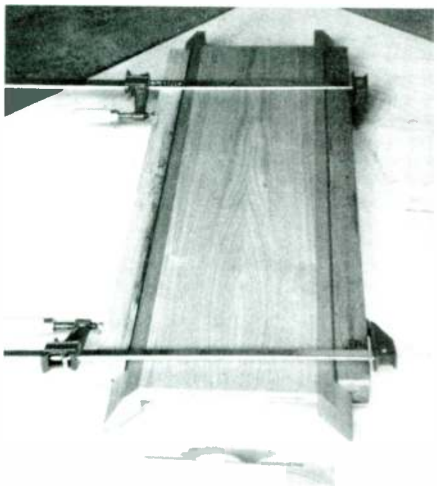

PHOTO 6: Assembly of front faceplate.

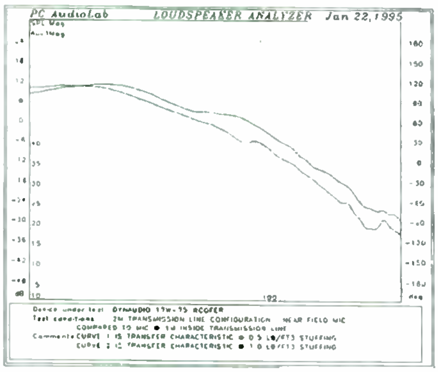

FIGURE 6: Frequency plot of 0.5 lb/ft^3 and 1.0 lb/ft^3 stuffed transmission line.

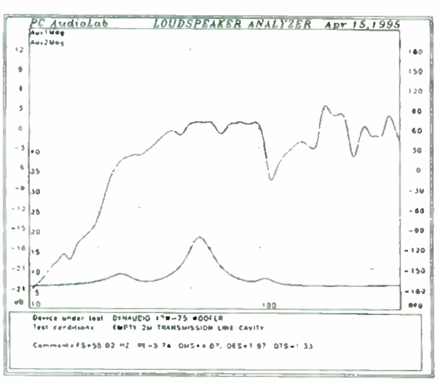

FIGURE 7: PC AudioLab impedance and SPL measurement result for unstuffed transmission line.

impulse and MLS analyzer, and a room-acoustics analyzer. And when you're not using the soundboard for testing electronics and loudspeakers, you can use it for computer games or as a CD-ROM multimedia sound device.

I used the loudspeaker analyzer to measure the Thiele-Small parameters of the Dynaudio 17W-75. The analyzer sweeps the synthesizer through a user-defined frequency range and measures the loudspeaker impedance magnitude and phase using a voltage-divider network. After it performed an impedance test, the analyzer automatically estimated the Thiele-Small parameters. You can also use this test setup to measure the driver mass M_{MD} with the delta-mass method, which involves adding a mass M_A to the cone and remeasuring the resonant frequency f_{SA}. You can then calculate the driver mass M_{MD} from:

$$M_{MD} = M_A/[(f_S/f_{SA})^2 - 1]$$

The driver-compliance C_{MS} and V_{AS} can then be calculated using:

$$C_{MS} = 1/[(2\pi f_S)^2 \times M_{MS}]$$

where

$$M_{MS} = M_{MD} + 0.575 \times S_D^{1.5}$$

and

$$V_{AS} = 1.42 \times 10^5 (S_D^2)(C_{MS}).$$

I used the measured values from *Table 1* in the TLBOXMODEL program to predict the TL low-frequency response.

ABSORBENT STUFFING

The TL cavity operates as a low-pass filter, in which the acoustic-line stuffing absorbs high-frequency components. The effective speed of sound also decreases as line-stuffing density increases. The PC AudioLab oscilloscope measured the change in the speed of sound in the damped line. I placed a microphone at the top of the line, which is approximately 3′ from the woofer. I set the synthesizer to generate a 100Hz tone burst lasting 50ms, and then measured the time delay between the electrical signal and acoustic response. Using this same measurement technique, the line was stuffed with 0.5 and 1.0 lb/ft^3 of Acousta-Stuf and compared with the time delay of the undamped line. *Figure 5* shows a plot of this tone-burst response.

The measured speed of sound with 0.5 lb/ft^3 of stuffing was 916 ft/sec, and with 1.0 lb/ft^3 was 782 ft/sec. For the sake of comparison, *The Loudspeaker Design Cookbook* uses the formula

$$c' = 1{,}130/[1 + 13.56 \cdot P]^{1/2} \text{ ft/sec}$$

where P = packing density of transmission-line stuffing (lb/ft^3).

Using this formula with the two stuffing rates, the predicted speed of sound would be 404 ft/sec and 296 ft/sec—not even close to the measured values, but the formula is for long-fiber wool, not the Acousta-Stuf and eggcrate foam we used. There may also be a discrepancy because our measured speed of sound used a 100Hz tone burst. A lower-frequency tone burst might show a different value for the speed of sound. The formula results aside, the TLBOXMODEL software most closely approximated the measured SPL curves using the measured speed of sound instead of the predicted values from this equation.

LOW-PASS MASS

To illustrate the low-pass filter characteristics of the stuffed TL cavity, I used two

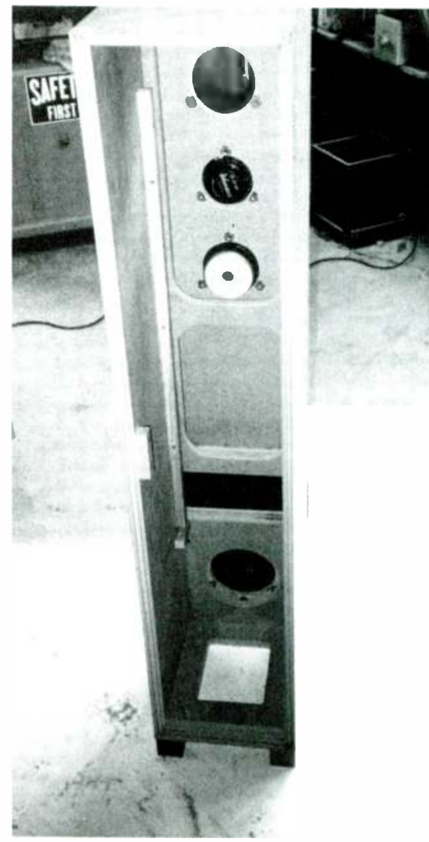

PHOTO 7: Rear view of main TL cabinet, with front faceplate attached.

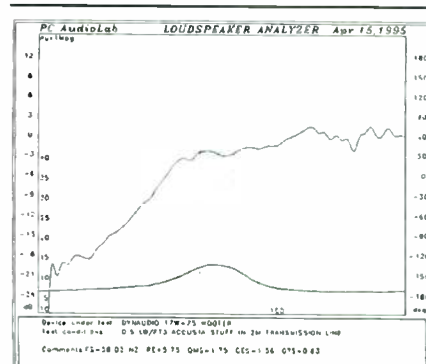

FIGURE 8: PC AudioLab impedance and SPL measurement for 0.5 lb/ft^3 stuffing.

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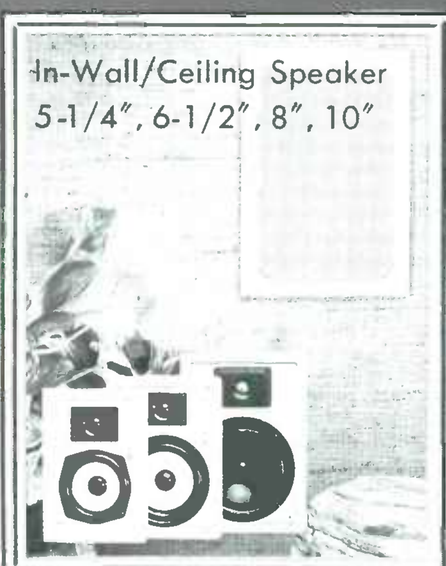

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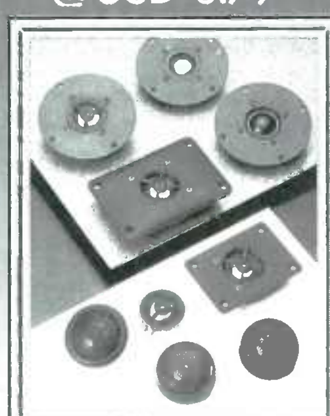

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microphones to compare the near-field response with that inside of the TL cavity. One microphone was outside of the cabinet next to the woofer diaphragm, while the other was inside the transmission-line stuffing, approximately 3′ from the woofer. *Figure 6* compares the TL-cavity SPL to the near-field SPL for stuffing densities of 0.5 and 1.0 lb/ft^3. From this graph you can see that the TL stuffed cavity acts as a low-pass filter, with a corner frequency of approximately 20–30Hz and an attenuation slope of 15dB/octave.

Figures 7–9 show the impedance and the near-field SPL of the TL woofer as varying amounts of acoustic stuffing were added to the line. I measured the SPL using a Mitey Mike placed 10cm equidistant from the port and speaker. *Figure 7* shows the undamped TL impedance and SPL measurement, *Fig. 8* the TL measurement with 0.5 lb/ft^3 of stuffing in the line, and *Fig. 9* the impedance measurement with 1.0 lb/ft^3 of stuffing in the line. Using the measured T/S parameters of the woofer and the measured change in the speed of sound in the stuffed TL cavity, the TLBOXMODEL software was reasonably close in predicting the low-frequency response of this system.

Part 2 of this article will present further construction details of the power supply, power amplifiers, and wiring, along with the design and measurement of the electronic crossover. It will also discuss measurement of the midrange, the tweeter, and the overall system response.

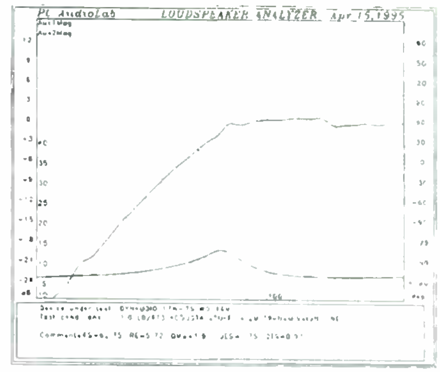

FIGURE 9: PC AudioLab impedance and SPL measurement for 1.0 lb/ft^3 stuffing.

PHOTO 8: Completed TL enclosure without base attached.

SOURCES

MicroAcoustics Audio Software Products
2553 Carpenter St., Thousand Oaks, CA 91362
(805) 495-8945, FAX (805) 373-5714
PC AudioLab v2.0SE

Old Colony Sound Lab
PO Box 243, Peterborough, NH 03458
(603) 924-6371, FAX (603) 924-9467
TLBOXMODEL, The Loudspeaker Design Cookbook

Part 3

THE WAVEGUIDE PATH TO DEEP BASS

By G.R. Koonce

This final section of my article describes the tests of some changes I made during the second listening session, as well as some follow-up work I tried with the experimental waveguide structure.

STIFF EXTENSION TEST

I had not tested the effects of stiffening the ¾L pipe extension before the listening session. The Waterfall plot (*Fig. 34*) for dual drivers X with the rubber throat and full-length stiffened extension shows some definite improvement in decay at low frequency as compared to the unstiffened extension (Fig. 32). I cut the extension in two during the listening session, and had not tested this configuration. *Figure 35* shows the NF response for dual drivers X with the rubber throat and stiffened 9″ length ¾L pipe extension.

This configuration comes closer to the proper 3:1 length ratio than the full extension, and the NF response does look smoother than the Fig. 31 results. I believed the full extension offered deeper bass in the listening tests, but the 9″ extension may be the better configuration. This again demonstrates the importance of maintaining the proper 3:1 ratio of the two pipe sections, even though the rubber throat seems to reduce the sensitivity to errors in this ratio in terms of developing an in-band dip.

Using the rubber throat with the correct 3:1 pipe-length ratio changed some of my conclusions about the rubber throat. With the full-length ¾L extension (Fig. 31), the above-band trash was quite a bit higher than the passband response, raising the question of possible high losses due to the rubber throat.

But *Fig. 35* clearly shows this effect was a function of the pipe-length error and not the rubber throat. The listening sessions made it clear that the rubber throat causes little loss. It's unfortunate I didn't recognize before the listening session that the 9″ extension configuration is the best configuration developed during this investigation.

ENERGY-TIME CURVES

Bill Waslo kindly took the data files for the time-response curves shown in Figs. 18 (unmodified) and 19 (stiffened) and developed energy-versus-time curves (*Fig. 36*, unmodified, and *Fig. 37*, stiffened) with his new Audiosuite software. Note that the waveguide configuration is dual drivers X tightly coupled to the normal length structure. These curves agree with the waterfall plots, Figs. 20 (unmodified) and 21 (modified); output after 50ms is reduced by the modifications.

ACTIVE EQUALIZER

Figure 38 shows the output from a program that allowed me to place an equalizer function ahead of the waveguide. The NF response of the waveguide is shown with and without equalization along with the equalizer response. I built an active equalizer based on this work (*Fig. 39*) and measured the response (*Fig. 40*). The waveguide really came alive and showed its low-frequency capability when I used this equalizer in a lis-

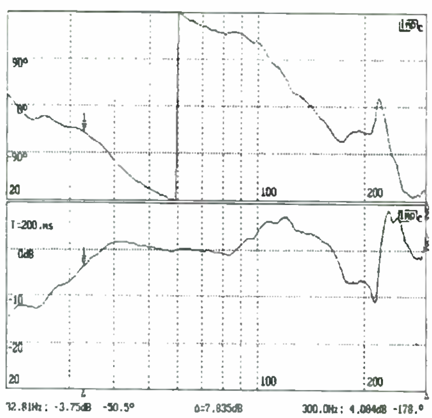

FIGURE 35: Near-field response for dual drivers X with rubber throat, stiffened 9″ ¾L extension.

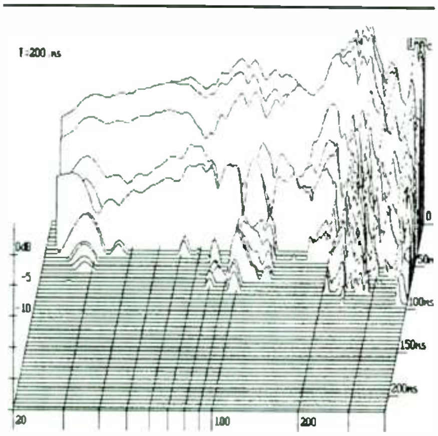

FIGURE 34: Waterfall plot for dual drivers X with rubber throat, stiffened full ¾L extension.

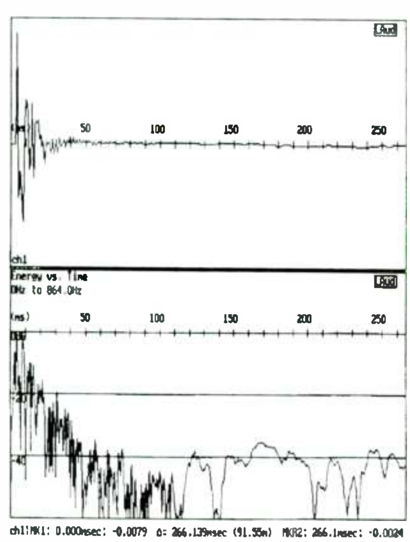

FIGURE 36: Energy-time curve for dual drivers X on original waveguide.

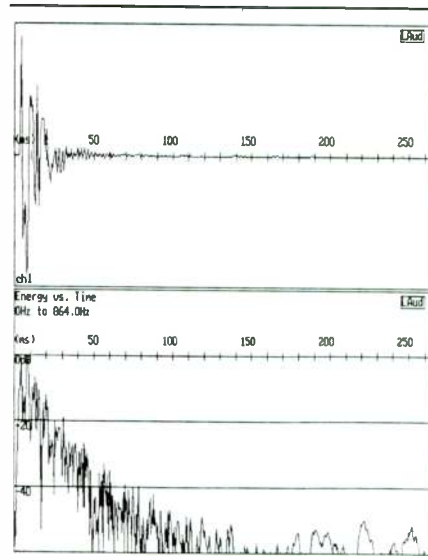

FIGURE 37: Energy-time curve for dual drivers X on modified waveguide.

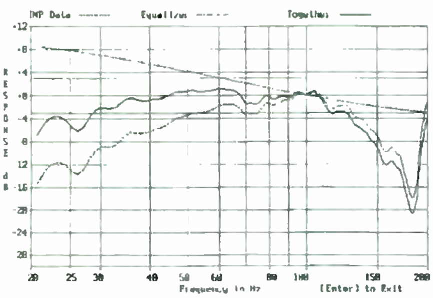

FIGURE 38: Computed performance of equalizer with waveguide.

tening session. Proper balance between the equalized waveguide and the high-end system was judged by ear at the waveguide padded 4–4.5dB below the upper-end system, consistent with earlier work, as the equalizer has 0.6dB gain at 100Hz.

I measured, via pink noise and the 1/3 octave RTA, the performance of the waveguide (*Fig. 41*) with dual drivers X, rubber throat, and full ¾L pipe extension driven through the active equalizer and active LP crossover (CO). The mike was back 2′ from each port, while the waveguide was driven at 2W, measured after the 110Hz second-order LP crossover and the equalizer.

The results do not appear flat down to 30Hz, as *Fig. 38* predicts, but this may be due to the third-order infrasonic filter in the system, which starts to offset the equalizer at low frequency. *Figure 42* shows the NF test results for the waveguide driven through the active equalizer (with no infrasonic filter) and indicates the response is within 3dB to about 25Hz. It is clear the experimental waveguide I built is at its best when used with equalization.

PASSIVE CROSSOVER & EQUALIZATION

There is no doubt active equalization and crossover is the correct method for the waveguide; however, I wished to determine if a passive approach was possible. The waveguide input impedance did not bode well for a passive approach, but I developed a program to place a four-branch passive network ahead of the waveguide. After some trial-and-error experimentation, I produced a network—consisting of a modified third-order 80Hz LP filter followed by a 40Hz Zobel—that did a respectable job.

Figure 43 is the predicted response of the network with the dual driver X with rubber throat and full-length extension waveguide; performance is good down to about 33Hz with the high-frequency trash well suppressed. *Figure 44* shows the input impedance to the waveguide and the predicted input impedance with the passive CO/equalizer. Any attempt to boost the low bass via this network naturally drove the network input impedance to a very low value.

Figure 45 is the network schematic, with both the design values and the actual component values used to implement the network. All capacitors were nonpolar electrolytics, and all coils except for the 45mH unit were ferrite-cored. The 45mH coil was four units in series wound on laminated iron-bar cores.

It is evident I did not meet the design Q values for the 5 and 12mH units. *Figures 46* and *47* measure the NF response and input impedance, respectively. Performance is nearly as predicted, except the high-frequency trash is not as well-suppressed, possibly due to the higher resistances of the 5 and 12mH inductors.

This work demonstrates that passive CO/equalization is possible, but at these frequencies the component values are tough to meet. Also, the dual 8Ω drivers in parallel mean that you are starting at 4Ω minimum impedance. The use of a passive crossover also requires matching the waveguide sensitivity to those of the other drivers in the system. Note that I did not use the waveguide with passive CO/equalizer in a listening session.

DRIVER DISPLACEMENT

This is still a major unknown that needs more work. As discussed earlier, with no equalization, driver displacement appears in general to double as frequency is halved. With a −6dB/octave equalizer acting over most of the waveguide passband, it tends to increase 4:1 as frequency is halved—equivalent to the performance of a normal direct-radiator system in its passband.

I currently have no idea of the true displacement function for the drivers in the waveguide. I have put 20W to 4Ω (8.94V) sine waves into the dual driver X waveguide over the 20–50Hz region with no sign of distress or damage. Computer programs indicate that I could drive these same levels into driver X in a B2 closed box, but the response at 20Hz would be down 28dB from the passband response. The computer indicates that in a QB3 vented box, 8.94V input at 20Hz would require driver X to have a maximum linear excursion of about 0.24″, which I doubt it has, even though the system response is down about 35dB.

Displacement function for the waveguide remains a mystery, at least to me. I currently do not know whether the waveguide input-power limit is set by the driver's thermal or displacement limit, and I certainly do not wish to run a test to find out.

DRIVER CONNECTION

With a vented or closed box system, I have always considered the performance deterio-

to page 34

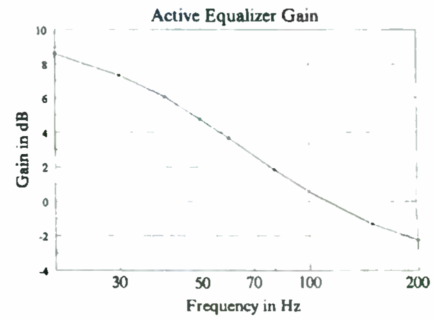

FIGURE 40: Measured response of active equalizer.

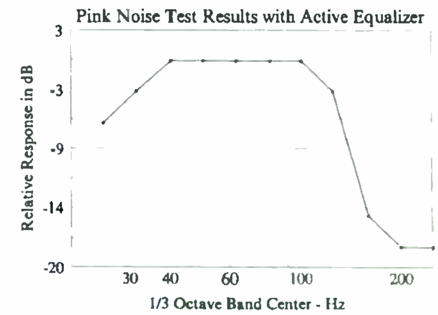

FIGURE 41: Pink-noise test results for waveguide with active equalizer and active crossover.

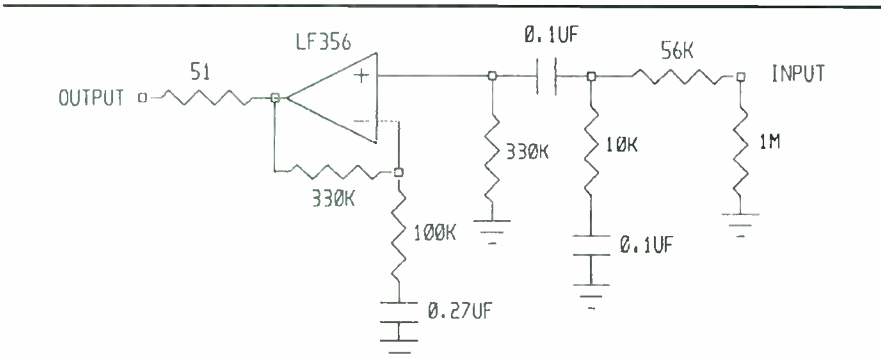

FIGURE 39: Schematic for active waveguide equalizer.

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Value µF	Diameter mm	Length mm	SRP US$	Value µF	Diameter mm	Length mm	SRP US$
1.0	11	21	1.23	12.0	25	33	3.56
1.5	12	22	1.44	15.0	25	38	4.18
1.8	13	22	1.49	20.0	29	38	5.16
2.2	15	22	1.58	24.0	29	43	5.98
2.7	14	25	1.67	30.0	32	43	7.30
3.0	15	25	1.73	33.0	32	48	7.74
3.3	16	25	1.78	41.0	35	48	9.32
3.9	16	25	1.83	50.0	37	53	10.96
4.7	18	27	1.96	51.0	37	53	11.16
5.6	18	30	2.10	56.0	39	53	12.00
6.0	19	30	2.20	62.0	39	53	12.98
6.8	20	30	2.33	75.0	43	58	15.12
8.0	20	33	2.91	82.0	45	58	16.28
8.2	21	33	2.97	91.0	47	58	17.50
9.1	22	33	3.08	100.0	49	58	18.76
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continued from page 31

rated when I switched from parallel to series connection on dual drivers. Test results indicate that even though the two drivers for the waveguide share a common small closed box, they tend to operate independently as the pipe load dominates. Connecting the drivers in series would certainly raise the low input impedance, so I tried it.

Figure 48 measures the input impedance with dual drivers X in series (still out of phase), with rubber throat and full ¾L pipe extension. This looks very much like an amplified version of the parallel driver Z_{IN} (Fig. 28). The NF response for series-connected drivers (*Fig. 49*) looks quite good except for the dip at 70Hz. The passband is extended at the high end, so it may be that series connection has some merit.

I repeated the test with the 9″ ¾L pipe extension, which produces close to the proper 3:1 length ratio, and a deeper dip appeared at 76Hz. These dips correspond to peaks in the impedance curves, indicating there is a potential problem with connecting drivers in series, beyond the loss in sensitivity. I did not use the waveguide with series driver connection in the listening sessions.

MODELING THE STRUCTURE

Bill Waslo[3] managed to model the dual drivers X with rubber throat and full ¾ pipe extension waveguide structure. In summary, he determined the following:

- The response curve is relatively insensitive to the driver T/S parameters.
- In terms of response shape, there seems to be an optimum pipe area for a given driver type, but it is not a function of driver cone area.
- Using a low Q_{ES}, low V_{AS} driver with a large pipe area can provide higher efficiency at the top end of the passband, but has little effect at low frequency.
- The waveguide structure is relatively inefficient, the dual drivers X structure being about 1.3% efficient at 100Hz. This would be 93dB/W/m radiating into 2π sr free-space.

The model findings agree in general with my test results. I did not test varying pipe area. The low efficiency the model predicted is a bit disappointing, but I believe it to be correct as developed below.

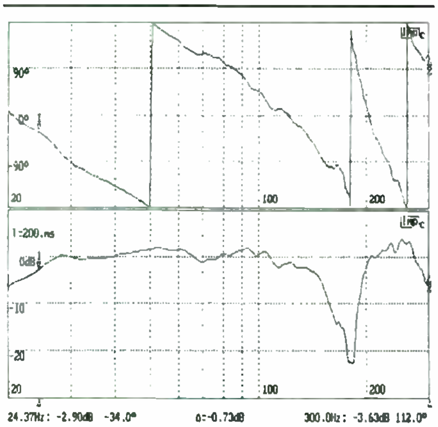


FIGURE 42: Near-field response dual drivers X, rubber throat, full ¾L extension, and active equalizer.

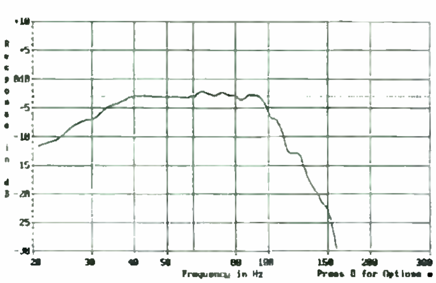


FIGURE 43: Computed NF performance of waveguide with passive crossover/equalizer.

SENSITIVITY, EFFICIENCY, ACOUSTIC GAIN

Sensitivity is the acoustic output from the system for a given input voltage, generally in terms of so many decibels for 2.83V input at a distance of 1m (dB/2.83V/m) radiating into 2π sr free-space. The reference normally used is that one acoustic-watt output produces about 112dB at 1m. Sensitivity is the other major unanswered question about the experimental waveguide system.

You're in for a shock if you think you can measure sensitivity by driving the system with a 2.83V sine wave and standing back 1m with an SPL meter. As Stuck and Temme[4] indicate, you must be sufficiently far back from the test system to ensure proper integration of all its outputs and also be in a true free-field measurement area, or you can't convert the result to a 1m reference. For a structure the size of the waveguide, this minimum distance is about 21′.

Even then, to use an SPL meter, you need an anechoic room good down to your desired test frequency—100Hz or below for the waveguide. In taking advantage of a time-windowed measurement, such as using a calibrated IMP, avoiding the first wall reflections would require a room about 32′ × 24′ × 24′! There was no way I was going to accomplish a true free-field measurement of sensitivity.

For those who have not tried such measurements at low frequency, *Fig. 50* indicates the problem, depicting the basic room layout and the waveguide location. The waveguide was driven with 8.94V (20W to 4Ω) sinusoids at 30, 40, and 50Hz, while I measured the SPL (C-weighted response, which is down about 6dB at 20Hz) at various points (#1–7) in the room, always at one-half room height.

Point #1 is 1m from both waveguide output ports. *Figure 51* plots the test results. Even though point #1 is the closest measurement point to the waveguide, it is never the highest SPL level in the room. The center of the room is generally low SPL, and the nearby corners are high SPL. It is clear you are measuring room-boundary effects and not waveguide sensitivity.

With piston drivers, Stuck and Temme indicate ways to make near-field measurements of sensitivity, which are virtually room-independent, and then convert the results to a 1m reference. I do not know a valid way to make this measurement with my equipment on a structure such as the waveguide, with its two side-by-side ports, and am open to suggestions from readers.

You can estimate sensitivity from the listening tests. The system used for the upper end has a computed sensitivity of about 88.4dB/2.83V/m. When used with the active equalizer and active crossover, the "match" between the waveguide and the upper-end

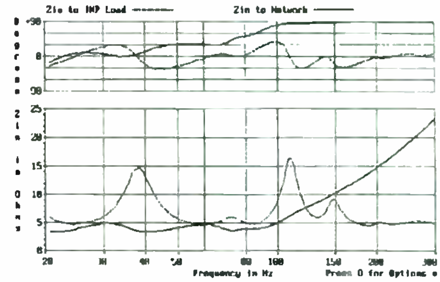
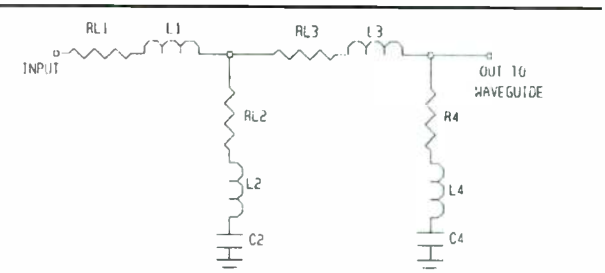


FIGURE 44: Computed ZIN to waveguide with passive crossover/equalizer.



	DESIGN	ACTUAL		DESIGN	ACTUAL
L1	12 MHY	11.83 MHY	L3	5.00 MHY	5.01 MHY
RL1	0.2 OHM	0.49 OHM	RL3	0.20 OHM	0.42 OHM
L2	0.8 MHY	0.81 MHY	L4	45.0 MHY	44.2 MHY
RL2	0.2 OHMS	0.25 OHMS	R4	8.0 OHMS	8.0 OHMS WITH RL4
C2	580 UFD	585 UFD	C4	300 UFD	299 UFD

FIGURE 45: Schematic for passive crossover/equalizer.

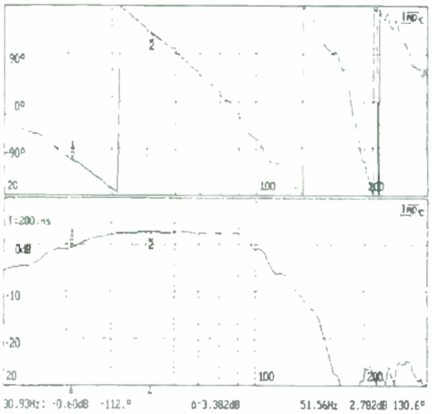


FIGURE 46: Near-field response for dual drivers X, rubber throat, full ¾L extension, and passive crossover/equalizer.

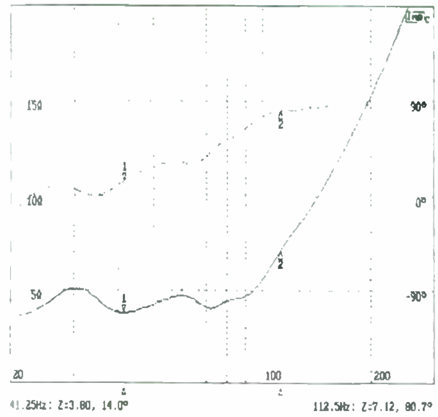


FIGURE 47: Z_{IN} for dual drivers X, rubber throat, full ¾L extension, and passive crossover/equalizer.

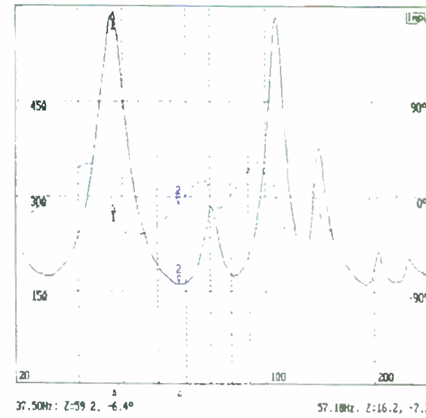


FIGURE 48: ZIN for dual drivers X in series with rubber throat and full ¾L extension.

system was set by ear always in the range of 4–4.5dB padding of the waveguide.

Combining this with the fact that the active equalizer has a gain of +0.6dB at 100Hz results in an estimated sensitivity of about 92dB/2.83V/m. Note that the computed sensitivity of each type X driver is about 93.8dB/2.83V/m in a direct-radiator application, but this work demonstrates that there is no direct relationship between a driver's sensitivity in a direct-radiator application and what it will show on the waveguide.

Taking the estimated sensitivity of 92dB/2.83V/m and using the actual input impedance for the waveguide at 100Hz yields a computed efficiency of about 1.5%, or 93.8dB/W/m, which agrees well with the model prediction. This is nearly the same as the reference efficiency (computed from T/S parameters) of 93.2dB/W/m for a single driver X. This low efficiency is also indicated by the input impedance curve (Fig. 28).

Structures with major acoustic gain, such as horns, produce a radiation resistance that raises the input impedance above the normal DC resistance. This occurs for portions of the waveguide Z_{IN} curve, but other portions are not far above the DC resistance.

While the experimental waveguide I built

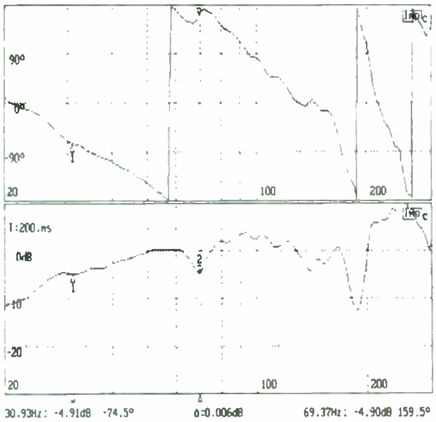


FIGURE 49: Near-field response for dual drivers X in series with rubber throat and full ¾L extension.

TABLE 6

RESPONSES (IN HERTZ)

Configuration	Expected Passband	IMP Indicated Passband	DB Spread Over PB
Close-coupled single-driver unit	28–141	35–145	9
Dual drivers, close coupled	36–180	34–220	9.5
Dual drivers, rubber throat (9" extension)	31–156	33–180	10
Dual drivers, rubber throat (full extension)	28-141	40–150	7

Notes: Passband has some ripple and peaking above 100Hz. Curves have slow rolloff at low end, so equalization could extend passband.

may show acoustic gain at its response peak, it yields little effective gain in the desired passband region below 100Hz. My waveguide structure produced reasonable bass levels in my garage down to the 25Hz range—certainly an accomplishment for two 8″ drivers. It will not produce the high acoustic levels normally associated with a horn system, and I cannot identify a passband gain anywhere near the 6dB with ripple that reference #2 (Part 1) led me to expect.

CONCLUSIONS

Based on the work to date, the following are my conclusions:

1. Rule #1: The entire structure must be *very* stiff.

2. Rule #2: For the waveguide configuration that I examined, it is very important to maintain the ¾L to ¼L length ratio at 3:1. If you miss by much, a dip will occur somewhere. The limited data seems to indicate that introduction of a rubber throat between the drivers and the waveguide reduced the magnitude of response dips generated by length-ratio inaccuracy, but maintaining the proper ratio is always preferable.

3. The crossover can be a problem. A lot of full-amplitude trash occurs above the passband. Also, the waveguide has wild impedance variations in the frequency range where you would like to do the crossover. It is likely that any system to which you cross over will also be going through impedance variations in this frequency range. While a passive CO/equalizer is possible, a far better approach is an active equalizer with a high-order active crossover and biamping.

4. The waveguide response is likely to need a −6dB/octave equalization over a portion or all of its passband.

5. Driver displacement is in question. I input lots of power into 6.5″ drivers in the

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continued from page 36

waveguide, which did not seem to hurt any of them. On one CD with heavy transient bass content, I heard the single 6.5" driver configuration overload in a way that did not occur with dual 8" drivers at an even higher signal level. With the single driver, you have, in effect, a driver with an open pipe hung on each side, and thus at some low frequency only the suspension keeps the driver cone in check.

In the dual-driver approach, the two drivers are mounted in a small closed box, but since they are driven out of phase, this box adds no stiffness, and at low frequency, all restoring force must again come from the two driver suspensions. With the dual 8" drivers, continuous sine-wave inputs of 10W per driver in the 20–50Hz range caused no problems. I listened with a system that has a third-order infrasonic filter at 18Hz. More work is certainly needed to gather information on the driver-displacement limits.

6. Sensitivity to driver T/S parameters: My waveguide-structure results were almost independent of the driver T/S parameters.

TABLE 7

EXTERNAL VOLUMES (AFTER STIFFENING) (IN FT³)

Structure	Volume
Bare waveguide	3.20
Waveguide with dual 6.5" driver box	3.52
Waveguide with dual 8" driver box	3.83
Waveguide with rubber throat and dual 8" box (no ¾L extension)	4.38
Waveguide with rubber throat and dual 8" box (full ¾L extension)	5.06
Waveguide with close-coupled single-driver box	4.10

There was no indication that the driver f_S or its reference efficiency correlated directly with what it did on the waveguide.

A single 6.5" 4Ω driver with a reference efficiency (computed from the T/S parameters) of about 85dB/W/m was able to fill the garage with bass below 110Hz. Two 8" drivers with f_S near 45Hz produced a waveguide system with useful output to at least 30Hz—even lower with equalization. Modeling work confirms the basic insensitivity to the driver T/S parameters.

7. Cone stiffness: This was the one driver factor that seemed to make a difference. My few drivers with stiff cones seemed to sound a little better. You could probably coat a fairly thin cone with something to stiffen it.

I did not try this, but I don't think it would hurt the efficiency as much as it does in a direct-radiator system. Certainly there is no need to maintain any high-frequency capability to any driver used in one of these low-frequency waveguide structures. It might help reduce the trash above the passband if you inhibit the ability of the driver to produce higher frequencies.

8. Waveguide pipe area: I used a very small waveguide area, intended for 6.5" drivers, and was able to obtain fairly good output levels. I performed no tests on varying the pipe area, and modeling indicated optimum pipe area was independent of driver-cone area.

You could probably improve overall waveguide efficiency at the top of the passband by using a larger pipe area and a low Q_{ES} and low V_{AS} driver. The example in reference #1 (Part 1) used a 12"-diameter driver (80% diameter area = 72.4 in²) and a waveguide area of 82.3 in². I have no other information on how to size this pipe area.

9. Coupling to the waveguide: I obtained best results with both cone faces tightly coupled to the waveguide (direct dual drivers), or with both cone faces coupled to the waveguide through a compliance (dual drivers with rubber throat). My single-dri-

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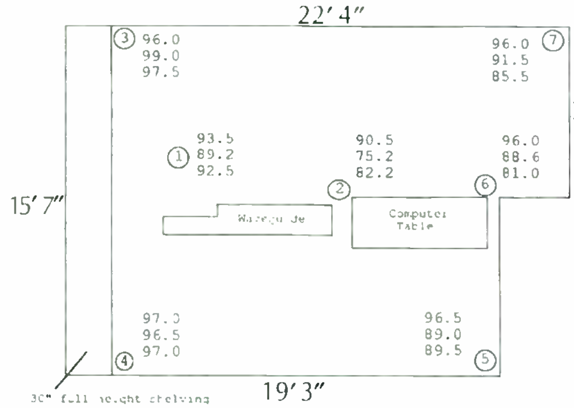

FIGURE 50: Basic layout of IMP testing room and SPL levels at various locations.

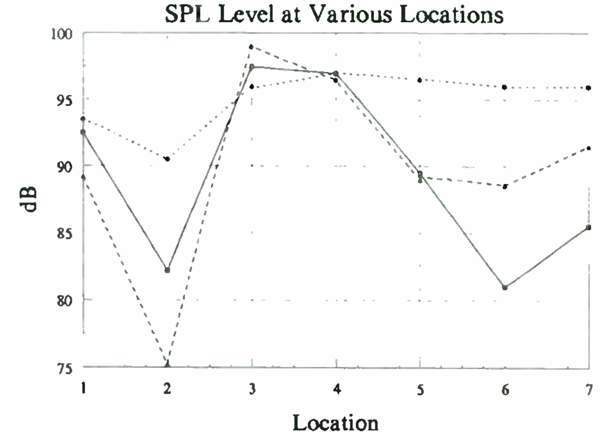

FIGURE 51: Plot of SPL levels measured in IMP test room.

ver approaches always had one cone face tightly coupled (front of cone) and the other coupled through a compliance; the bass extension here was not as good for a given pipe length. While I did not test this, it is possible to build with a single driver having compliance volumes on both faces. The rubber-throat approach seems more tolerant of length-ratio error, but is definitely best with the correct length ratio, and may cause a small loss of sensitivity.

10. Waveguide length versus passband: Table 4 (Part 2) lists the lengths and length-ratio accuracies of the various waveguide configurations I tested, as nearly as I could determine. *Table 6* relates these lengths to what you might expect the passband to be and what IMP NF testing has indicated. Per reference #2, the expected passband should be about λ/4 to 5λ/4 of the ¾L portion of the waveguide, or, if the waveguide length ratio is correct, from λ = 3L to λ = 3L/5.

I converted these values to frequency and listed them in *Table 6* as the expected passband. The IMP indicated passband results come from taking the upper and lower −3dB points from the NF response as best as I could read them. The decibel-spread value is from the −3dB level to the top of the peak above 100Hz.

The dual close-coupled drivers perform about as predicted. The single drivers in the close-coupled single-driver unit produce a longer waveguide, which should extend the low end. Tests do not indicate this extension, but the results vary somewhat from driver to driver, and the low-end cutoff slope is very gradual, so it is difficult to define the low-frequency cutoff point.

Remember also that the single driver is really not coupled as tightly on the back side of the cone as on the front, or as tightly as the dual-driver cones. The dual drivers with the rubber throat seem to perform about as predicted when the pipe-length ratio is correct (9″ ¾L pipe extension) and seems a bit more tolerant to error in the length ratio.

It would thus appear that for both sides of the waveguide tightly coupled, or both sides of the waveguide coupled through compliances, the basic equations of reference #2 *do* predict the passband, which is not flat but generally rises with frequency and has a peak at the upper end (Fig. 2). Remember also that considerable high-frequency trash exists above the passband for the crossover to deal with. For design purposes, if the lowest usable frequency is F_L in hertz, then, according to reference #2:

Upper end of the passband = 5 F_L hertz.

The total pipe length L = v/(3 F_L), where v is sound velocity.

L = 4,512/F_L in inches for v = 1,128 ft/s.

11. I sized my rubber-throat volumes by guess. Perhaps a review of horn references would offer some insight here.

12. Sizing the dual-driver chamber: I tried to make this chamber as small as practical in an attempt to "force" Isobarik operation. Tests have shown operation is not Isobarik with the waveguide pipes controlling the driver, even with the rubber throat in use. While I didn't test it, I suspect basic operation would be the same if the box were left open, except for the rear radiation adding to the system output, which means that in effect you have two single-ended waveguides driven out of phase.

Individual closed boxes for each driver might also be acceptable, as that would definitely provide protection for the drivers in terms of excessive displacement with very low-frequency input. The cover of this chamber was a major source of acoustic leakage, and filling the box or boxes with damping material might help. I currently have no scientific way to size this chamber.

13. Efficiency/sensitivity: While my experimental waveguide appears to offer gain at the response peak, little or no effective gain was indicated in the passband I preferred to use. That does not mean that the waveguide structure can't produce higher outputs of low frequency than the same drivers could do in a direct-radiator application; whether or not this is true depends on the unknown displacement function, which is usually the limiting factor for small drivers at low frequency.

It does mean, however, that a small driver with a low electrical-power limit will not produce the high acoustic outputs that it could in a horn application. The size of the waveguide for perhaps 25Hz reproduction is, however, much smaller than that of a horn; *Table 7* lists the external volumes for the various waveguide configurations for comparison purposes. With dual 8″ drivers X, the rubber throat, and full-length ¾L extension, the sensitivity of my waveguide at 100Hz was estimated at 92dB/2.83V/m, and the efficiency at 93.8dB/W/m.

14. Testing: If you plan to do serious work with one of these experimental waveguide structures, you need some acoustic test method. Because of the narrow dips caused by improper length ratios in the waveguide, the pink noise and RTA were not ideal for waveguide testing. IMP is perfect for this sort of work, and you should consider it if you do not currently have a good method.

SUMMARY

This three-part series investigates an experimental, double-ended waveguide structure intended to produce bass from about 35Hz to over 100Hz with 6.5″ or 8″ drivers. Such

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REFERENCES

3. Personal communication with Bill Waslo.

4. C.J. Stuck and S.F. Temme, "Simulated Free Field Measurements," *JAES*, Vol. 42, No. 6 (June 1994): 467–482.

SOFTWARE SUPERSTARS FROM DOWN UNDA

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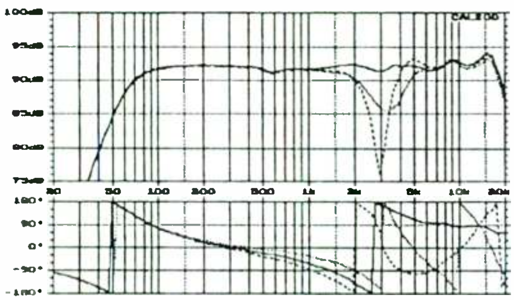

CALSOD 3.00o response of a two-way system calculated for the on-axis design case, as well as +/-15° above and below the principal listening axis.

CALSOD

CALSOD (Computer-Aided Loudspeaker System Optimization and Design), developed in Australia by Witold Waldman of Audiosoft, has for many years been one of the most extensive, most respected, and most dependable audio software packages in the world. Although written for PC/AT/286/386/486/Pentium and PS/2 under MS-DOS, its excellent CGA/EGA/VGA/Hercules graphics support mimics Windows in many ways.A math coprocessor is not required, but will be used if present; 512K RAM needed. CALSOD is available in both Standard 1.30o (hard disk not required) and Professional 3.00o (hard disk required) versions (current update is lower case "oh"). A 240- or 370-page User's Manual is supplied on disk, with accompanying hard copy illustrations, and further support, if needed, is always available from Old Colony.

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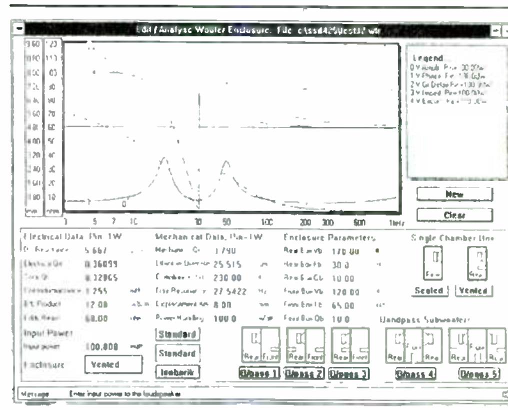
The software functions performed by SSD+ are as amazing to see as they are easy to use. And contrary to a misprint in some recent publicity, NO spreadsheet software of any kind is required to run SSD+. Ten different modules (Editor; Box; Box Optimization; Compensation; Crossover; Sum Plot; Filter Optimization; Crossover Optimization; Box Time; Time Res) operate from 10-40,000Hz using 520 discrete frequency points. The vertical amplitude scale covers 65-125dB, while phase is displayed within +/-180 degrees.

SSD+'s capabilities include speaker driver library creation; enclosure design for seven types—single chamber sealed and vented, sealed-vented, 2 types of vented-vented, sealed-vented-sealed, and triple vented; enclosure optimization; compensation of driver impedance or amplitude; crossover design and optimization; system frequency response evaluation and optimization; system in-room frequency response; calculators; thermal analysis; polar plots; time domain responses; and importing data from IMP, LMS, and CLIO.

"We found SSD to be a very interesting and useful package...It (is) quite easy to (use), and operators will appreciate the common Windows environment..." —Rob Evans, Technical Editor, *Electronics Australia*.



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Wayland's Wood World

SCREWS AND OTHER FASTENERS

By Bob Wayland

I'll begin by admitting to a prejudice: the use of screws in speaker construction is a sign of poor craftsmanship. I know this is not fair; there are places where fasteners are necessary. How else can we hold in our drivers or attach the terminal box? And we all like to take the back off and fiddle around with what is inside. I draw the line, however, at using screws in making joints.

Applying a first-class finish over a screw head, even if carefully plugged, is difficult. Of course, you can make the covered screw hole a part of the enclosure design. I will restrict this discussion to the speaker-building aspects of fasteners, and the tricks of using them correctly.

TYPES

The simplest classification of screw types is based on their head shapes. The most common—flat and round—are shown in *Photo 1*. The seldom-used oval head completes the formal list, but there are also specialty heads (*Photo 2*) and types (*Photo 3*). The parts of a screw include the shank, thread, and core, as shown in *Fig. 1*.

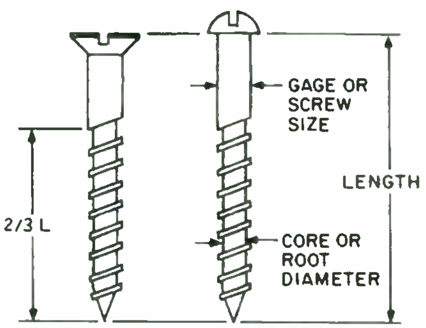

FIGURE 1: The parts of a screw.

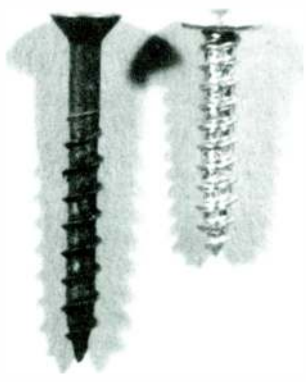

PHOTO 1: Two common screws: on the left is a common flathead, with threads starting about one-third of the way down the shank; on the right is a roundhead, with tapping threads the full length of the shank.

Because you normally want a flush surface, the flathead screw is best for solid wood. Tapping screws, which have threads the full length of the shank, are designed for particleboard (PB) and medium-density fiberboard (MDF); they also have deep threads and smaller core or root diameters. You can go to a lot of trouble to obtain special tapping screws for PB and MDF, but I find that drywall and deck screws more than meet my needs at less expense.

The most important factor affecting ease of use is the head style, which determines the type of screwdriver the screw will accept (*Photo 2*). The best by far is the square drive, followed by the Phillips head.

LOAD RESISTANCE

The screws you use in speaker building need to withstand the pressure exerted in securing one element to another. If the screw doesn't fail in tension, its effectiveness depends on how well it resists withdrawal from the wood. *The Encyclopedia of Wood* has a lot to say about the Forest Products Laboratory's research on withdrawal, or load, resistance (the force required to pull the screw out without unscrewing it).

To summarize Forest Products' study, the formula for the withdrawal load of a wood screw inserted into the side grain of seasoned wood is

$$p = 15,700\ G^2DL$$

where p is the withdrawal load in pounds, G is the specific gravity of the wood, D is the shank diameter of the screw in inches, and L is the length of penetration (in inches) of the threaded part of the screw.

This study raises a number of points. First,

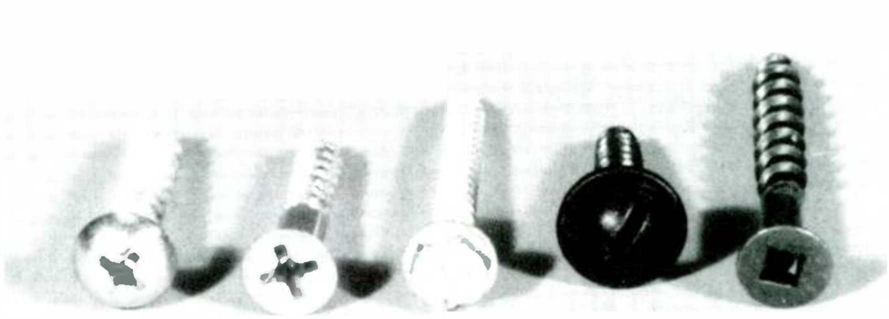

PHOTO 2: Drive configurations. From the left: two Phillips, a socket drive with a slot or straight drive, a slot with built-in washer, and a square drive.

for hardwoods with high specific gravities, such as walnut and oak, the withdrawal resistance may exceed the tension-failure limit, making the screw the weak link. Also, for heavy hardwoods—maple or purpleheart, for example—you gain little by using longer screws.

Sheet-metal, or tapping, screws have about 10% greater withdrawal resistance than other types of the same size, assuming that the pilot hole is about 70% of the screw thread's root diameter for softwoods, or about 90% for hardwoods. Going into end grain without splitting the wood, the resistance is about three-fourths of that for side grain. (By the way, using soap or glue—or, even better, beeswax—to facilitate insertion has little effect on the withdrawal resistance.)

One consideration the study doesn't mention is that a tapered drill produces a pilot hole that profiles the shape of wood screws, thus increasing their load resistance.

Everything changes, of course, when you work with PB and MDF. The withdrawal resistance then becomes:

$$p = KD^{1/2}(L - D/3)^{5/4}G^2$$

where K is 2,655 for PB-face withdrawal load, but 2,055 for screws in the edge of the board. The other variables are the same as in the seasoned-wood formula. The kicker here is that you must tighten the screws to 60–90% of the stripping torque, T, which for 8-gauge screws at a depth of 5/8″ is given by:

$$T = 27.98 + 1.36X$$

The torque is in inch-pounds, and X is the density in pounds/cubic feet. Thus, moderate

PHOTO 3: Hanger bolt and roundhead screw.

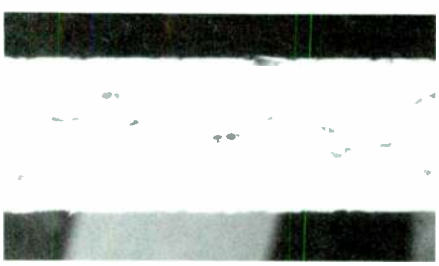

PHOTO 4: Cross section of PB. Notice the hard, dense surfaces and the less dense center.

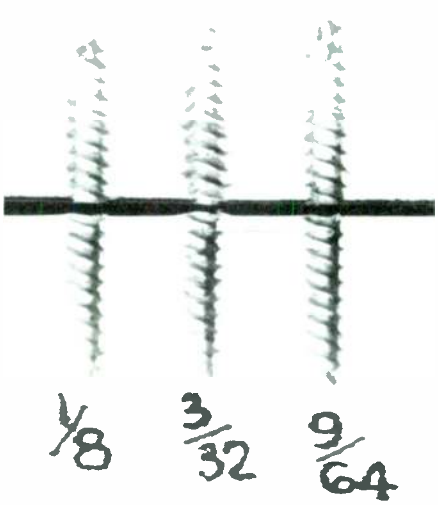

PHOTO 5: A study of pilot-hole size in forming good penetration patterns for screws.

tightening produces the best results. One advantage of using MDF is that K increases to 3,700 for the face and 2,860 for the edge.

Most PB faces are dense, but the interior is less so (*Photo 4*). For this reason, it is not a good idea to screw into the edge of PB.

PRACTICAL CONSIDERATIONS

The woodworker often overlooks, to his or her disadvantage, two facets of using screws: the correct size of the pilot hole and its proper preparation. Regarding the former, builders often try to select drill size according to root diameter by sighting across the threads. Aside from the obvious parallax problem, this almost always results in a pilot hole that's too large. It should be a bit smaller than the screw's root diameter (70% for softwoods and about 90% for hardwoods). The tapered drills designed specifically for a fixed-size screw are not optimal, but a compromise. The suggested pilot-hole diameters listed in *Table 1* differentiate between softwood and hardwood.

You can perform a simple test that demonstrates what happens to the holding ability of a screw. Cut a piece of the wood you are

TABLE 1

BIT SIZES (IN INCHES)

SCREW SIZE	0	1	2	3	4	5	6	7	8	9	10
SHANK HOLE	1/16	5/64	3/32	7/64	7/64	1/8	9/64	5/32	11/64	3/16	3/16
PILOT HOLE SOFTWOOD	1/64	1/32	1/32	3/64	3/64	1/16	1/16	1/16	5/64	5/64	3/32
PILOT HOLE HARDWOOD	1/32	1/32	3/64	1/16	1/16	5/64	5/64	3/32	3/32	7/64	7/64

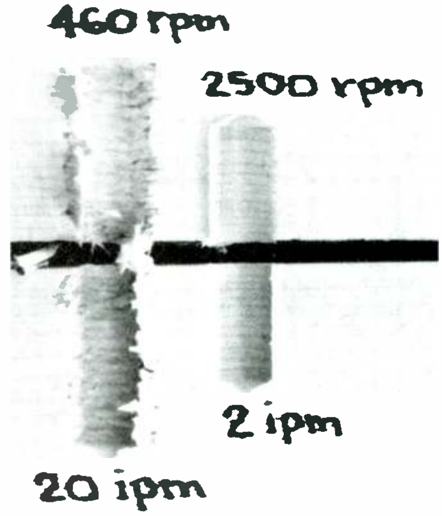

PHOTO 6: Effects of drill RPM and feed rates on pilot holes. A larger drill was used to help illustrate the effect.

PHOTO 7: Installing a hanger bolt.

using in half with a fine-toothed saw blade. Now tightly clamp the two pieces together end to end and drill a series of holes along the cut: one the size of the recommended pilot hole, another that's 1/32″ over, and a third that's wider by 3/64″. Now insert and remove the screw from each hole in turn. When you unfold the cut, the results should be similar to those shown in *Photo 5* (for a #10 screw in white pine).

With the correct-size hole, the threads penetrate fully, and the wood fibers are cleanly cut and pulled around the pilot hole. With the 1/32″ oversized hole, the threads don't fully penetrate into the sides of the pilot hole, and so will not provide maximum holding power. The hole that's 3/64″ larger produces an unacceptable holding pattern. In this soft wood, the 1/32″ oversized hole is acceptable, but when you are working with hardwoods, be careful to have the correct-size pilot hole, as an undersized hole could split the wood.

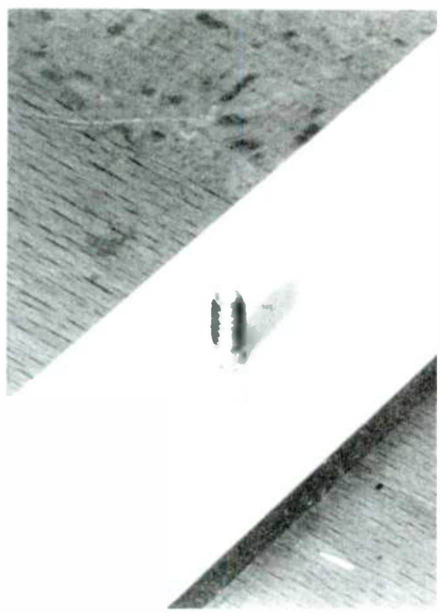

PHOTO 8: Installed hanger bolt.

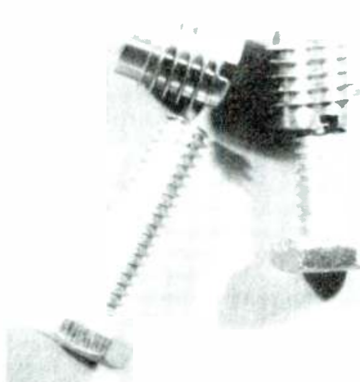

PHOTO 9: Solid-brass inserts and bolts.

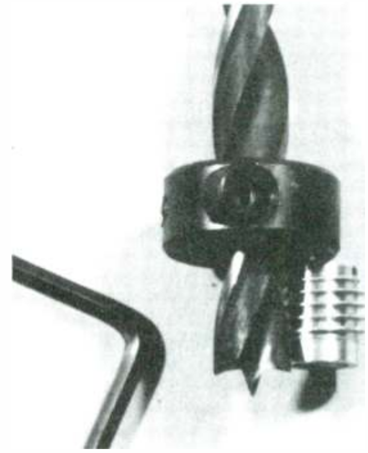

PHOTO 10: Setting the depth of a drill stop for an insert.

As always, the devil is in the details.

The other important consideration with pilot holes is how carefully you drill them. Using the same setup, drill two holes. When drilling the first one across the cut, use a low RPM and a fast feed rate (e.g., 460 RPM feed at 20 ipm). For the second hole, use a high RPM/slow feed rate (e.g., 2,500 RPM feed at 2 ipm). *Photo 6* shows the results for white pine. Clearly, you would not expect good holding from the low-RPM/fast-feed hole! So always use the highest RPM and lowest feed rate reasonable for the operation.

REUSABLE FASTENERS

Whether you're developing a new speaker system, tuning, or just doing routine maintenance, the ability to remove the back panel is important. This process can quickly tear up the screw holes and leave a most unsatisfac-

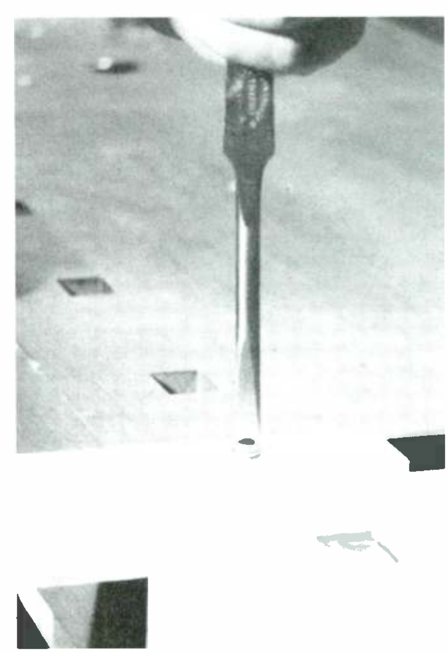

PHOTO 11: Screwing in an insert. Be sure to use as large a screwdriver as possible.

PHOTO 12: Installed insert. Note that it is just below the surface.

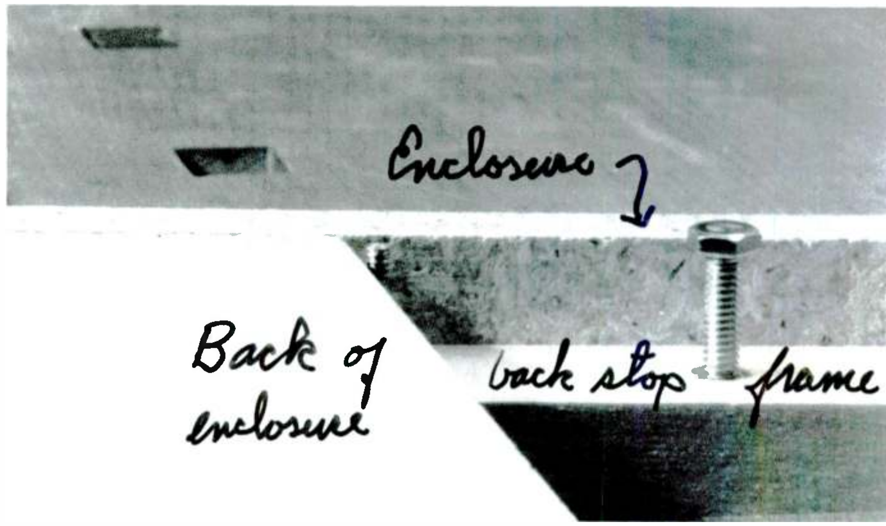

PHOTO 13: Relationship of the enclosure back, backstop frame, and the enclosure.

tory mess. A simple solution, if you plan it in advance, allows you to remove the back as often as necessary, with no deterioration of the fastening system.

The procedure utilizes hanger bolts, such as those shown in *Photo 3*. Keeping in mind the above discussion, prepare the pilot hole. When it is ready, place two nuts on the bolt part of the hanger, locked against each other as in *Photo 7*, and use a wrench to insert the hanger bolt to the full length of the screw (*Photo 8*).

Space the bolts closely enough to ensure an air-tight seal. (If you are gluing the back, there's no problem, but for our experimental enclosures the seal can be tricky.) The key is the elastic property of the back material. Not surprisingly, this depends upon grain direction, as well as species, moisture content, and curing techniques. When you use manufactured boards, such as PB and MDF, the situation is somewhat better because of the uniformity of the product.

INSERTING THE BOLTS

First, you need to distribute the load around the bolt hole. As with the pilot holes, drill the bolt holes at high RPM and low feed rate. A smooth hole allows uniform distribution of the compression from the bolt. If the hole is too small, the board will split when you tighten the bolt. Make the hole just large enough so you can insert the bolt by tapping it lightly with a wooden mallet.

The bolt hole should not be too close to the edge of the board. As a rule of thumb, keep it at least 1.5 times the board's thickness away from the edge. A good general practice is to make a frame inside the cabinet that is recessed away from the cabinet's back face. Make this backstop frame separately and then put it in position. The frame, fitted inside the cabinet and recessed at least the thickness of the back material, must be thick enough to allow for the offset of holes in the back.

To prevent bolt heads from protruding beyond the back of the enclosure, be sure the backstop frame is sufficiently recessed from the back edge of the enclosure sides so that the bolts are below the plane established by the enclosure's back edge. Of course, you

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Product Review

AUDIOCONTROL C-101 SERIES III OCTAVE EQUALIZER AND REAL-TIME SPECTRUM ANALYZER

By Gary A. Galo
Contributing Editor

AudioControl, 22410 70th Ave. W. Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166. Cost: $459.

AudioControl is well known to *SB* readers for its affordable equalization and loudspeaker/room analysis equipment. In *SB* 2/87 (p. 50) I reviewed AudioControl's Richter Scale Series III, which combined a two-way, fourth-order electronic crossover with a low-frequency warble-tone analyzer and a five-band low-frequency equalizer. The warble-tone analyzer was one of the best I had seen, and I still use it for low-frequency in-room measurements.

The C-101 Series III combines a two-channel, ten-band graphic equalizer with a wideband pink-noise generator and real-time analyzer (*Photo 1*). The equalizer section consists of ten faders spaced at one-octave intervals, from 32Hz up to 16kHz. Unlike some graphic equalizers, the C-101 has the left- and right-channel controls placed side by side. For example, the first two faders are 32Hz, left and right. This is logical, since left- and right-channel adjustments are likely to be similar in most cases.

In a normal stereo system, you should connect the C-101 to one of the tape-monitor loops on your receiver, integrated amplifier, or preamp. AudioControl has built a tape-monitor loop into the C-101, to replace the one it uses. In addition to the 20 faders, the front panel has four other switches associated with the equalizer section. The Equalizer switch, when depressed, activates the equalizer circuitry. If this button is out, the signal passes through the input and output audio circuitry, but not through the equalizer.

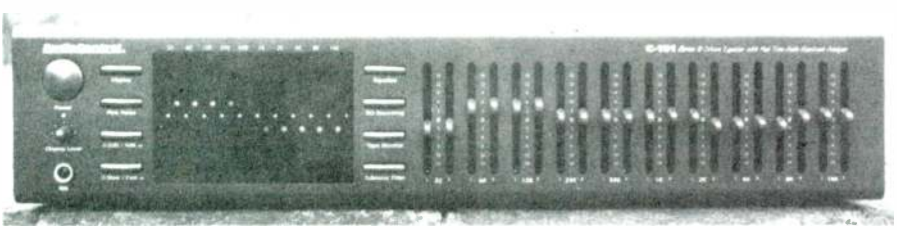

PHOTO 1: The C-101 Series III equalizer.

An EQ Recording switch allows you to send an equalized signal to your tape deck. The Tape Monitor switch activates the C-101's tape-monitor loop, and functions exactly like its counterpart on your preamp. Finally, the Subsonic Filter switch engages an 18dB/octave filter at a corner frequency of 20Hz, which is handy for minimizing woofer excursions caused by warped LPs. This switch should really be called an Infrasonic Filter. I took AudioControl to task for using this incorrect terminology back in my review of the Richter Scale, but my advice went unheeded.

The analyzer section is remarkably straightforward. The front-panel display consists of a line of small yellow LEDs across the center, representing a 0dB reference point. A rectangular matrix of red LEDs displays the levels above or below the 0dB reference at each of the ten center frequencies. AudioControl supplies an omnidirectional electret condenser microphone with the C-101. Electret condenser mikes have proven themselves to be ideal for room and loudspeaker analysis. The mike plugs directly into the front panel of the C-101.

THE PINK-NOISE GENERATOR

The C-101's triple-source pink-noise generator is combined with averaging circuitry to improve the accuracy of the output. AudioControl is correct in calling this a "lab grade" generator. Depressing the Pink Noise switch engages the pink-noise generator, and the front panel displays the spectrum picked up by the microphone. There's a level-control pot on the rear panel for adjusting the generator's output level. A front-panel-display level control allows you to move the entire display up or down. For example, you might wish to adjust the level so the 1kHz display reads 0dB before you begin equalizing your system.

When the pink-noise generator is not engaged, the front panel displays a spectrum of the program material you're playing. You can switch off the display if you don't wish to follow the bouncing LEDs. You can also switch the display so each increment represents either 4dB (for coarse adjustments) or 2dB (for fine tuning). Furthermore, you can adjust the analyzer's response time with a Slow/Fast switch. The slow position is generally used for making equalization adjustments, while the fast position is best for monitoring program material.

Adjusting your room/loudspeaker interface is extremely easy. After centering the display as I mentioned above, simply move each slider on the equalizer until the corresponding band on the display reads 0dB. Some interaction will occur between the bands, but it doesn't take long to get the hang of it. If you are adjusting the equalizer for personal taste, rather than a flat response, the analyzer still allows you to see the effects of your adjustments.

AudioControl supplies an excellent 50-page manual, which makes an already easy-to-use product even simpler. The manual is not your typical, stuffy user's guide written by an engineering team. It is informal, down-to-earth, and often humorous. At the same time, it is extremely accurate and easy to understand. I can't imagine any user having difficulty operating the C-101 after reading this manual. If you've used a graphic equalizer before, operation is probably self-explanatory.

The C-101 manual gives a thorough overview of the equalization procedure, complete with many helpful photographs. The manual also offers a review of basic acoustical theory, so users not familiar with terms such as "octave," "frequency," and "hertz" should have no trouble understanding the terminology. AudioControl also offers excellent instructions for connecting the C-101 to your system, covering a variety of connection schemes, including hooking up the C-101 with other signal processors.

TECHNICAL DETAILS

Although AudioControl didn't supply a schematic with the C-101, a look inside reveals the general circuit-building blocks (*Photo 2*). The equalizer and audio circuitry are built around 23 JRC 4560 dual op amps. AudioControl refers to these ICs as "state-of-the-art high-speed integrated circuits." The 4560 is competitive with most dual op amps, but a number of single IC op amps from Analog Devices, Linear Technology, and Burr-Brown outperform the 4560. C-101 doesn't use conventional signal switching. Instead, all switching is done with CMOS ICs to keep the signal paths as short as possible and minimize the possibility of noise pickup.

Power-supply regulation consists of two 7815 and two 7915 fixed, three-terminal IC regulators. On the heels of Walt Jung's power-supply series in *Audio Amateur* 1/95 and 2/95, cheap three-terminal regulators are a truly "low-end" way to build power supplies. I'd like more manufacturers of low-cost products to use three-terminal adjustable regulators, such as the Linear Technology LT-1085/1033 pair. These cost very little more to implement than the 7815/7915 regulators, and their performance is significantly better.

AudioControl uses trifurcated sliders in the equalizer, which it claims are the most reliable type for this application. All RCA jacks are gold-plated, PC-mount types. Unlike some PC-mount jacks, those AudioControl uses actually have a gold-plated center sleeve. Many products contain jacks with gold-plated shields, yet with tin-plated center conductors, as if gold is unnecessary where it can't be seen. I congratulate AudioControl for avoiding the P.T. Barnum approach to RCA jack selection.

Elsewhere, the parts quality is not up to high-end audio standards. All resistors in the audio circuitry, including the equalizer circuits, are 5% carbon film. Inputs and outputs are capacitor-coupled with 22µF electrolytics, and AudioControl uses no high-frequency bypassing across the electrolytic caps. The equalizer circuits contain many Mylar® capacitors, and there are even four ceramic disc caps in the 8kHz and 16kHz sections.

LISTENING TESTS

Obviously, the C-101 is an effective ten-band equalizer/analyzer that performs exactly as advertised, so far as frequency-response analysis and adjustments are concerned. But I was also interested in evaluating its sonic performance, independent of frequency-response alterations. I set all controls to their center (flat) position with the front panel equalizer switch depressed. I used CDs exclusively during my evaluations.

I discovered that the C-101 minimized many of the refinements I had worked so hard to achieve in my system. It reduced the soundstage size, both left to right and front to back, and localization of instruments was less precise. On the Decca/London opera recordings produced by John Culshaw, including Solti's Wagner *Ring*, I found it more difficult to determine where Culshaw had placed the singers in the soundstage. Subtle stage movements were sometimes lost.

My system has an excellent sense of air and space around the instruments, well demonstrated with delicate recordings such as Ravel's *Alborada del Gracioso*, conducted by Ernest Ansermet (London 433 717-2). With the C-101 in the signal path, however, the sense of air and spaciousness in the high frequencies virtually disappeared, and the pristine detail in this recording became somewhat diffused.

On Antal Dorati's Mercury recording of Respighi's *The Birds* (432 007-2), the C-101 imparted an edge to the violin tone. This is a difficult recording to reproduce, but if everything in your system is up to par, the string sound is clean and free of hardness. The

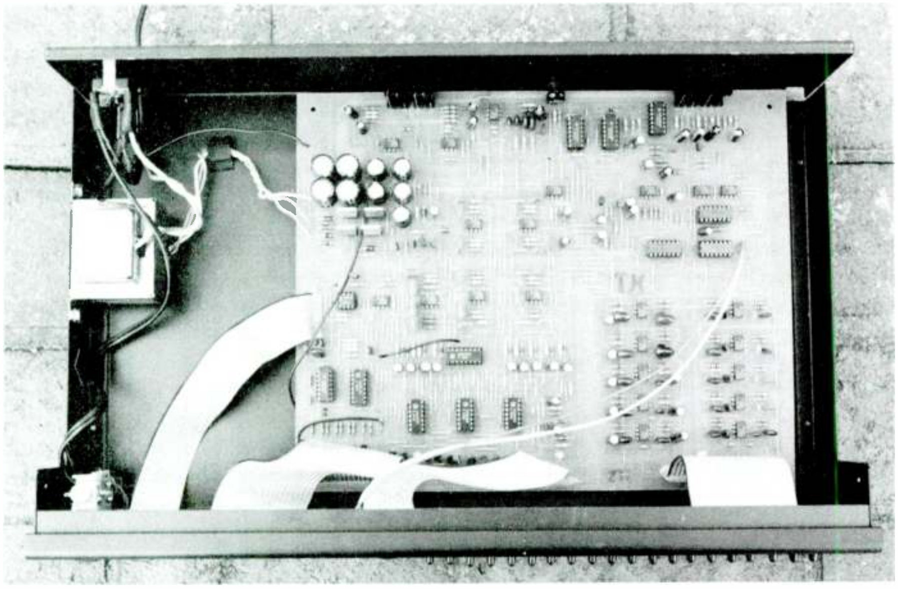

PHOTO 2: The C-101's circuit-building blocks.

C-101 also tends to reduce dynamic contrasts. It lessens the weight and impact of heavily scored passages, and soft passages do not seem to descend effortlessly into the noise floor in the way to which I am accustomed. On the other hand, the C-101 did not add any audible hiss or hum to my system, even at high playback levels.

WHO NEEDS IT?

One school of thought in audio says, "frequency response fixes everything." Get the frequency response flat, or adjusted to your personal taste, and all those expensive high-end circuits, tweaks, and modifications become unnecessary. Competently designed audio circuits have no sound of their own, except for alterations in frequency response, phase response, and overall level. So you can add as many circuits as you need to adjust the response to your requirements without otherwise affecting the sound.

I am not a subscriber to this philosophy. I believe in minimalist signal paths in an audio system—refined, high-resolution circuitry with high-quality parts, *but the less of it, the better*. My preamp doesn't even have tone controls. A list of associated equipment appeared in my review of Audio Concepts' Sapphire III loudspeakers (*SB* 3/95, p. 40). It hasn't really changed since then, so I won't repeat it here. The system may look complex, but the actual analog signal path is a purist, no-frills approach.

Like any graphic equalizer, the C-101 puts a proliferation of circuitry in your signal path. Whether you need this depends on the rest of your system. If you own a receiver from Asia and loudspeakers of modest capabilities, you may find the C-101 a useful addition to your system. But if you have a true high-end audio system, any equalizer, short of a 'Cello *Audio Pallette* at $15,000, may pose as many sonic problems as it solves.

Your taste in music may also affect your decision to purchase an equalizer. If you prefer classical music, and your goal is to reproduce the sound captured by the recording engineer as accurately as possible, you probably won't care to change the tonal balance of the recordings. AudioControl says (manual, p. 39) that "no recording is 'correct' as you hear it. It has simply been equalized the way the producer or artist desired...." Also, recording engineers use equalizers on every channel of their mixing consoles, and CD mastering engineers add "even more equalization to 'balance' the various songs so they have similar tonal characteristics."

I'm sure that many classical producers and engineers might take issue with these statements. The best classical recordings usually involve simple microphone techniques and no equalization in the recording process. I'm certain Wilma Cozart would be horrified if a CD mastering engineer added "more equalization" to her carefully prepared mixdowns of Mercury's extraordinary three-track master tapes.

According to AudioControl: "If you like tons of bass, order some up. If a vocal is indistinct, enhance it. If you just want to fiddle around, fiddle around! That's what the recording producer and engineer did. And now it's your turn as well." These suggestions may appeal to rock-music fans, but I believe they are inappropriate for classical-music listeners. Of course, even classical-music listeners sometimes must alter a particularly bad recording. A recording with audible tape hiss or line-frequency hum can sometimes be made more palatable with judicious equalization. But, a parametric equalizer is better suited to this task than a graphic.

LIVE PA

Equalization is usually necessary in live sound-reinforcement applications. Unfortunately, most affordable equalizers have limited sonic performance and don't include a real-time analyzer. The C-101, however, is one of the best low-cost, ten-band equalizers I've seen. It doesn't add appreciable noise to the system, and it is clean enough for any sound-reinforcement application. My audiophile criteria to judge the C-101 for critical home listening aren't relevant to this task.

Since the C-101 has unbalanced RCA input and output jacks designed to interface with home-audio line levels, I wouldn't recommend using it with expensive pro-audio mixers with XLR connectors operating at +4dBm. It works extremely well with cost-effective PA mixers that have unbalanced ¼" phone plugs operating at nominal levels of −2 or −10dBm. I highly recommend the Mackie mixers for cost-effective sound reinforcement. They pack a lot of performance into affordable products, and will work well with accessories such as the C-101.

As AudioControl notes in its manual, the C-101 is built for light-duty home use. It wasn't constructed to be kicked around in the back of a truck. But most pro-audio gear built to withstand abusive road use is also extremely expensive. If you use the C-101 for sound reinforcement, treat it with care. I'm putting my money where my mouth is as far as this application is concerned, for I have purchased the review sample for sound-reinforcement use at the Crane School of Music, SUNY Potsdam.

BENCH TESTS

I measured total harmonic distortion (THD) and SMPTE intermodulation distortion (IM) using a Sound Technology 1700B Distortion Analyzer, which measures SMPTE IM distortion using 60Hz and 7kHz at a 4:1 ratio. The C-101 equalizer, with controls set flat, is a unity-gain device. I used a 2V input signal, which results in a 2V output. I first measured THD with the front-panel equalizer switch out, the "subsonic" filter off, and the Sound Technology's 80kHz low-pass filter in. In the left channel, 1kHz THD measured 0.0054%, and 0.005% in the right. With the 80kHz low-pass filter out, distortion measured 0.022% in the left channel and 0.019% in the right. The distortion products were dominated by second and third harmonics, plus noise.

At 10kHz, with the 80kHz low-pass filter in, the left channel measured 0.0058% and the right 0.0055%. Without the 80kHz filter, the left measured 0.0022% and the right 0.018%. Some high-order distortion products were present in both cases. Measured distortion levels at both 1kHz and 10kHz actually dropped very slightly with the equalizer switched in, but high-order distortion products visibly increased at 10kHz. SMPTE IM distortion measured 0.018% with the equalizer switched out, and dropped to 0.013% with it switched in. The left and right channels were identical.

AudioControl specifies the noise level at −118dB, referenced to full output, which is 8V RMS (my measurements showed clipping at 8.7V RMS). This figure isn't particularly useful, since you'll probably never operate the equalizer at full output. I measured noise relative to 2V RMS out. With the equalizer switched out and the input shorted, left-channel noise measured −76dB and the right channel −79dB. The noise levels varied no more than 1dB with the inputs open, indicating that they are largely unaffected by source impedance. Switching the equalizer in raised the left-channel noise level 2dB and the right 5dB.

Relative to 8V, my measurements showed noise levels of −87dB in the left channel and −91dB in the right, with inputs shorted and the equalizer switched out. These are substantially lower than AudioControl's figures, but my measurements are unweighted. Although AudioControl doesn't specify, I suspect its measurement is weighted, which would account for the difference. These noise figures are certainly good for a product with so much active circuitry.

With the equalizer switched in, the frequency response was 1.5dB down at 100kHz and 0.5dB down at 10Hz. With the EQ switched out, the response was flat to 100kHz and 0.25dB down at 10Hz. I made these measurements with the "subsonic" filter off. With the filter on, I verified AudioControl's specification of an 18dB/octave Chebyshev alignment at 20Hz.

SUMMARY

The C-101 Series III is a good, cost-effective equalizer/analyzer that should perform well in any situation requiring an equalizer. Whether or not the C-101 is the best solution to your room/loudspeaker irregularities will depend on the rest of your system and the type of music you prefer. If you have a true high-end system, you will be better off working on your room acoustics and loudspeakers. In modest, receiver-based systems, however, the C-101's sonic artifacts aren't likely to be noticed.

MANUFACTURER'S SPECIFICATIONS

- **Total Harmonic Distortion**: 0.005%
- **Frequency Response**: 10Hz–100kHz ±0.75dB
- **Noise**: –118dB (Ref. full output)
- **Maximum Input**: 8V RMS
- **Maximum Output**: 8V RMS
- **Input Impedance**: 100kΩ
- **Output Impedance**: 150Ω
- **Control Bandwidth** ("Q"): 2.5
- **Control Center Points**: 32, 60, 120, 250, 500, 1k, 2k, 4k, 8k, 16kHz
- **Control Range**: ±15dB
- **Subsonic Filter**: 20Hz, 18dB/octave Chebyshev alignment
- **Power Consumption**: 10W
- **Dimensions**: 17" W × 3.5" H × 11" D
- **Weight**: 9 lbs
- **Country of Origin**: US

MANUFACTURER'S RESPONSE

First, we thank Gary for purchasing the C-101 that he reviewed. Second, we appreciate the comments on the power supply and suggestion about the regulation. This is a new regulation design we probably will use in the future, but, unfortunately, the articles in Audio Amateur *were printed after this unit was manufactured.*

With regard to some of the parts used, and specifically the 5% resistors, we have a suggestion for all readers. That suggestion is, simply, to measure. With the quality control in most factories these days, 5% resistors are more like 2% resistors (with the large majority within 1%). Commonly, when we measure parts, they are much better than their published specs. This tends to be true with parts which have been in production for a while, but with older printed specifications. It is not unusual for us to find actual slew rates more than twice those on ten-year-old spec sheets. Also, in the case of AudioControl, most of our components are inserted by automatic

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Software Review

ELECTRONICS WORKBENCH

Reviewed by Edward J. Aho

Electronics Workbench, Windows and DOS versions, $299. Interactive Image Technologies Ltd., 908 Niagara Falls Blvd. #068, North Tonawanda, NY 14120-2060, (416) 361-0333, FAX (416) 977-1818.

Circuit design can be most rewarding; it offers the opportunity to tap into creative and inventive skills we too often let atrophy. At times, though, you can become "stuck," either in the mathematics or in selecting the best circuit topology for a particular application. You can also spend hours designing, testing, modifying, redesigning, and retesting the circuit until it works and can be optimized. But help is available, in the form of a powerful program that not only enables you to perform schematic capture and PCB design, but also to simulate electronic circuits on your personal computer, thereby performing nondestructive experiments that accelerate your design process.

Interactive Image Technologies Ltd. offers Electronics Workbench, with which you can design circuits on the fly. By entering a schematic into the computer, you can simulate the circuit's operation without leaving your desk, trying different circuit topologies quickly and safely without the bother or expense of stocking a large array of components. The program lets you change, remove, or add components until the circuit is optimized, so when you move to the lab bench, you can do so with confidence and the expectation of success.

WHY THIS SOFTWARE?

To be honest, there are other, much more powerful electronic-circuit simulation packages that will perform such methods as Monte Carlo and worst case analysis; however, when price and ease of use are important considerations, Electronics Workbench is the leader of the pack. And don't get me wrong. This software is by no means a slouch; it offers fast, reliable information. It also includes a digital, as well as an analog, module, which makes the price of this package that much more attractive. Most other manufacturers sell these modules as separate packages, and at much steeper prices.

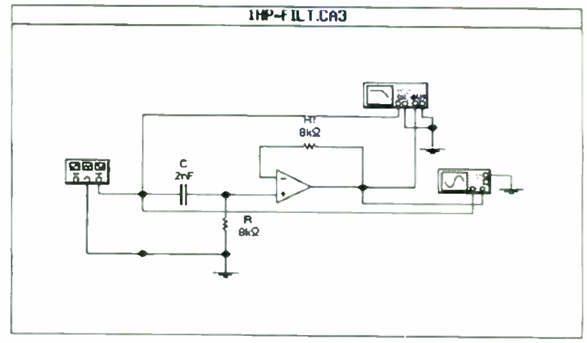

FIGURE 1: Sample of an Electronics Workbench circuit simulation.

Electronics Workbench has a remarkably shallow learning curve. You can achieve meaningful results just minutes after installing this program. And, if you are just learning electronics theory, or are interested in advancing your understanding, it can be an invaluable tool. It allows you to try what you like without the smoke and other excitement the same experiments can produce in the lab, and it offers almost instant results, which you can print out for later examination and review.

OBTAINING RELIABLE RESULTS

The capability of any software to simulate an electronics circuit depends entirely on how well the circuit is modeled. Those who have used this type of software know just how meaningful that statement is, but for the newcomer I'll explain. Let me start by asking, "When is an inductor just an inductor?" The answer is "never!" At low frequencies, the resistance of the wire that makes up the coils of the inductor can be a factor requiring careful consideration, and at high frequencies the capacitance between each coil may be the critical factor.

Keep in mind that perfect or ideal components do not exist: the way a component is fabricated can be a cause for concern in certain applications. You must consider the characteristics of your components, and then decide how you can model them so they can be properly analyzed. For example, you can model a real-world inductor by placing a capacitor in parallel with it and a resistor in series to simulate its characteristics.

Now, let's assume you have considered the inductance of your wirewound resistors, the ESR (Equivalent Series Resistance) of your capacitors, and so on; but how about those nonlinear components such as the diode, transistor, and op amp? Have you thought about your layout? Is it an important consideration? What about the inductance in the long leads to the gates of FETs?

What all this means is that you, the circuit designer, must know your circuit. You will

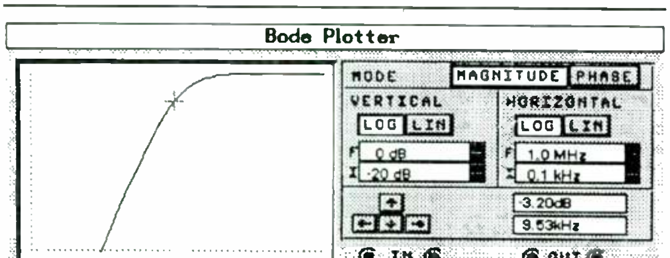

FIGURE 2: The Bode plotter display.

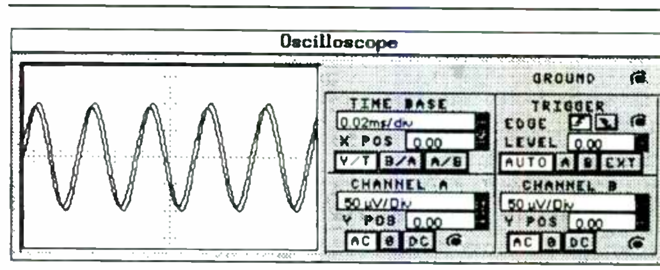

FIGURE 3: The oscilloscope display.

need to decide when the ideal component will suffice, or when a model of the real-world component is necessary to obtain a reliable simulation. Now, don't fret—you will most likely start off by considering *all* your components ideal, and then later on model the components you deem necessary.

Electronics Workbench makes this an easy transition by providing both ideal and real-world models for transistors, diodes, and op amps. Changing from one to the other is an easy operation—you simply click twice on the desired component and a window pops up allowing you to make the changes by modifying the appropriate boxes.

A CLOSER LOOK

The installation is simple. When the program executes, you will be looking at a large workspace, a parts bin containing a wide array of components, and a shelf containing fully functional lab instruments such as an oscilloscope, a Bode plotter, a function generator, and a multimeter. This shelf also contains the power switch that starts each simulation. You can analyze either the transient or steady-state response by selecting the appropriate one.

Building and testing a circuit is simple and fast. By using your mouse, you can drag components and test equipment from the parts bin and instrument shelf to place them on the workspace. They are connected by clicking the mouse on one terminal, dragging the wire to the next terminal, and releasing the button.

The software routes the wire automatically. If you don't like the routing, you can move the components to change it. This automatic wire placement speeds up the process, but it can be annoying for a perfectionist who prefers a schematic to look a certain way.

You can change component values by pointing the mouse at the desired component and clicking the button twice, which calls up a window containing the component's value. You label components by using the pull-down menus.

SAMPLE SIMULATION

The sample simulation (*Fig. 1*) illustrates what this software can do and the type of output it provides. Unfortunately, the graphical results of the analysis leave much to be desired. I set up the simulation to provide a Bode plot (*Fig. 2*), but this graph cannot be enlarged, either on screen or when printed, for better inspection. Also, the vertical and horizontal axes are poorly labeled, resulting in a difficult-to-read graph.

The saving grace of this display is the crosshairs, whose coordinates appear in the Bode plotter (lower-right corner). You can drag the crosshairs to any point in the display, which is handy for finding the cutoff frequency, or any other significant point. Unfortunately, cursors are not provided for the oscilloscope simulation (*Fig. 3*).

CONCLUSION

I enjoyed working with this software, which provided fast results and was simple to learn, although it did crash on occasion. When this happened, it displayed an error message I did not understand, closed itself, and then sent me back to my Windows logo. That was quite frustrating, but it's my only real complaint.

Electronics Workbench can be an exceptional buy for the student, hobbyist, technophile, or anyone on a budget who does not need the more sophisticated analysis the higher-priced simulation packages provide. For those who perform circuit analysis only occasionally, it can be a huge time-saver, which, of course, usually translates into saving money, as well.

DEVELOPER'S RESPONSE

We appreciate the chance to comment on this review and to update readers about what we believe is an exciting new product.

Since this review was written, Interactive Image Technology has released significant enhancements that address many of the issues in the review. For example, Electronics Workbench now gives users the ability to simulate analog, digital, and mixed-mode circuits and to produce larger and more complex circuits. Also, the new version's zoom scope feature improves the labeling and readability of the graphical results. The graph now can "be enlarged, either on screen or when printed, for better inspection."

Other new features include:

- *more analog and digital parts*
- *over 300 additional analog models*
- *real-world digital models (TTL, CMOS, TP, Buffer or OC)*
- *A/D and D/A converters*
- *timer, monostable multivibrator*
- *SCR, triac, diac, Schottky diode*
- *variable resistor, capacitor, inductor*
- *encoder, decoder, multiplexer, demultiplexer*
- *74xx, 74xxx ICs*
- *"record" components for saving simulation results to ASCII file*
- *improved user interface for easier wiring*

IIT's latest version is 4.1, which is Windows 95 and NT-compatible and 32-bit. Price is $299, with the Engineer's Pack costing $599. Version 4 owners will receive a free upgrade; the upgrade for earlier-version owners is $79.

Roy Bryant
Vice President & General Manager

SB *Mailbox*

CLOSE RECALL

Clarence C. Close points out that in his article "Rebuilding the KLH-9 Power Supply" (*SB* 2/95, p. 24), although the text correctly refers to the diodes in the updated power supply as 1N4007s, the schematic of Fig. 1 indicates they are 1N4005s. The text is correct.

FORMULAS FOR SUCCESS

While browsing through *Bullock on Boxes* and *The Loudspeaker Design Cookbook* (4th edition), I noticed that Mr. Bullock's formula (p. 6):

$$Q_{TS} = Q_{ES}'Q_{MS}/(Q_{ES}' + Q_{MS})$$

differs from Mr. Dickason's (p. 132):

$$Q_{TS}' = Q_{ES}'Q_{MS}/(Q_{ES} + Q_{MS})$$

Which of these is correct?

Also, with regard to Bill Waslo's V_{AS} formula (*SB* 4/93, p. 39):

$$V_{AS} = \frac{0.0327D^4[(f_S/f_{SM})^2 - 1]}{f_S^2 M_A}$$

How is the constant 0.0327 derived?

Roman Lock
Tustin, CA 92680

Contributing Editor Robert Bullock responds:

Mr. Dickason's formula is incorrect. The second Q_{ES} should also be primed. Otherwise, you are using two different values for the same parameter.

The coefficient in the V_{AS} formula comes from conversion of the meter-kilogram/second formula

$$V_{AS} = \frac{\rho_O c^2}{4}\,\frac{D^4[(f_S/f_{SM})^2 - 1]}{f_S^2 M_A}$$

into a foot-ounces/second formula.

Contributing Editor Bill Waslo responds:

The constant 0.0327 comes about as follows:

Small gives V_{AS} (the volume of air having the same acoustic compliance as the driver) as equal to

$$\rho_O \times c^2 \times C_{AS}$$

where C_{AS} is the acoustic compliance of the driver suspension and $\rho_O \times c^2$ is the density of air times the square of the velocity of sound in air, or 142,500kg/m/sec².

C_{AS} expressed in mechanical terms is $C_{MS} \times S_D^2$, where C_{MS} is the mechanical compliance of the driver suspension and S_D is the effective radiating surface area of the driver. If the effective diameter of the driver is D, then using the formula for the area of a circle, S_D is:

$$S_D = \pi \times (D/2)^2$$

In the mass method described in my article, the value C_{MS} is determined by how the resonant frequency of the driver changes when mass is added to the cone. If the mechanical mass is called M_{MS} and the added mass is M_A, then the resonant frequency of the mass-compliance combination is given by

$$f_S = 1/(2 \times p \times \sqrt{(C_{MS} \times M_{MS})})$$

for the driver alone and

$$f_{SM} = 1/(2 \times \pi \times \sqrt{(C_{MS} \times (M_{MS} \times M_A))})$$

for the driver with added mass. With some algebra and good strong coffee you can then extract from the above:

$$C_{MS} = (f_S^2/f_{SM}^2 - 1)/(f_S^2 \times M_A \times (2 \times \pi)^2)$$

If we bring this all together, we get

$$V_{AS} = \rho_O \times c^2 \times (f_S^2/f_{SM}^2 - 1)/(f_S^2 \times M_A \times (2 \times \pi)^2) \times (\pi \times (D/2)^2)^2$$

Pull all the constants together and you get:

$$V_{AS} = [\rho_O \times c^2 \times (\pi/4)^2/(2 \times \pi)^2] \times D^4 \times (f_S^2/f_{SM}^2 - 1)/(f_S^2 \times M_A)$$

The constant in brackets equals 142,500kg/m/s²/64 = 2,226.5kg/m/s².

Now, just to make sure that everything is really nice and confusing, we have to be able to enter the value for D in inches (for ease of use by US readers) and get the result in ft³ (the measure mostly used as input by box-modeling programs). Hence we need to multiply by a factor of $(1/39.37)^4$ to get the D input converted from inches into meters, and by a factor of $(3.28083)^3$ to get the V_{AS} output from m³ to ft³. This then gives a constant of

$$(1/39.37)^4 \times (3.28083)^3 \times 2{,}226.5 = 0.0327$$

Pretty obvious, right? (Just kidding.) The above assumes that M_A enters in kilograms. You may need some other conversion factors if you will be using other units of measure for D or M_A, or need the result in other terms than ft³. Liberty Audiosuite and IMP both accept units in either grams or ounces, inches or centimeters, cubic feet or liters.

QUESTIONS ANSWERED

I will attempt to answer some of Steve Pleasant's questions (*SB* 6/95, p. 58). Peerless makes a superb midrange (821385) and tweeter (811815) for moderate power applications. In general, European woofers do not have a heavy enough sound for Mahler or organ music. When the Peerless midrange and tweeter are combined with Madisound's Swan 305 woofer, you get the best of both worlds—finely detailed treble with solid bass that is not boomy. A 115-liter ported cabinet will yield an F_3 of 28Hz. This woofer moves more air per dollar than any other I know of. If this is too large, consider using the Madisound 1052DVC woofer in a 63-liter ported cabinet for an F_3 of 35Hz. At low volume, this woofer will actually sound even cleaner than the Swan. There is no need for a subwoofer with these combinations.

A close friend of mine has done extensive work with Dynaudio components and various 12- to 18-inch woofers. At high volume levels, his systems are markedly superior to his ears and mine, but we are talking about levels that get apartment dwellers evicted. If you like your music really loud, Dynaudio midranges and tweeters are your only choice. Parallel Swans would be an inexpensive woofer possibility worth trying with the Dynaudio mid and tweeter.

Cone materials affect the timbre of a speaker by the harmonics and modulated noise they produce. The level of this noise is too low to affect the response curve, but is always audible. To my ears, soft domes impart less coloration than metal. In general, polypropylene woofers have better low midrange than paper-cone woofers, since they permit a higher crossover frequency. This is a controversial aspect of speaker work; it is art, not science. How you hear these timbre effects is a very individual matter. Many listeners perceive metal domes to be brighter than the curve would predict.

Balancing speaker components requires a good ear and unending patience. I never use third-octave measurements, because I find the speaker systems that result from using them to be rather bright. Although my friend has access to Bruel, Kjaer, Audio Precision, and everything else a university has to offer, he still uses his ears for final analysis. I feel there is a rational approach to driver selection using elementary school math.

Jesse W. Knight
Woburn, MA

AUTHOR CORRECTIONS

There were a couple of errors in my article "Computer-Aided Bass Horn Design" (*SB* 5/95). The data as printed in Fig. 2 (p. 19) is incorrect. See *Table 1* for correct data.

TABLE 1

DATA FOR A HORN DESIGN

Driver's parameters: Name	Peavey 12"
Fs	50.4 hertz
Qes	.246
Fhm	409.7561 hertz
Vas	5857.92 cu. inches
Horn data: Flare rate	50 hertz
M	.6
Type	Corner Horn
Rear chamber	2047.76 cu. inches
Mouth area	725.1496 sq. inches
Throat area	33.80299 sq. inches

LENGTH Fm THROAT	HORN AREA AT X	ONE-HALF HORN AREA	5 INCHES A SIDES	24 INCHES B SIDES
2	35.79323	17.89661	7.158646	1.491385
4	38.00002	19.00001	7.600004	1.583334
6	40.4425	20.22125	8.088499	1.685104
8	43.14184	21.57092	8.628367	1.797577
10	46.12145	23.06073	9.224291	1.921727
12	49.40717	24.70358	9.881433	2.058632
14	53.02748	26.51374	10.6055	2.209478
16	57.01377	28.50689	11.40275	2.375574
18	61.40061	30.70031	12.28012	2.558359
20	66.22603	33.11301	13.24521	2.759418
22	71.53187	35.76593	14.30637	2.980494
24	77.36413	38.68206	15.47283	3.223505
26	83.77338	41.88669	16.75468	3.490558
28	90.8152	45.4076	18.16304	3.783967
30	98.55063	49.27531	19.71012	4.106276
32	107.0467	53.52337	21.40935	4.460281
34	116.3772	58.18861	23.27544	4.849051
36	126.6229	63.31147	25.32459	5.275956
38	137.8727	68.93637	27.57455	5.744698
40	150.2242	75.11208	30.04483	6.25934
42	163.7843	81.89216	32.75686	6.824347
44	178.6708	89.33539	35.73415	7.444616
46	195.0126	97.50628	39.00251	8.125524
48	212.9514	106.4757	42.59028	8.872975
50	232.6428	116.3214	46.52857	9.693452
52	254.2576	127.1288	50.85152	10.59407
54	277.9831	138.9915	55.59661	11.58263
56	304.025	152.0125	60.805	12.66771
58	332.6092	166.3046	66.52184	13.85872
60	363.9834	181.9917	72.79668	15.16598
62	398.4197	199.2099	79.68394	16.60082
64	436.2168	218.1084	87.24335	18.1757
66	477.7022	238.8511	95.54044	19.90426
68	523.2356	261.6178	104.6471	21.80148
70	573.212	286.606	114.6424	23.88383
72	628.0645	314.0323	125.6129	26.16936
74	688.2689	344.1345	137.6538	28.67787
76	754.347	377.1735	150.8694	31.43112

Also, under the breakhead "PROGRAM MODS" (at second bullet), X = X = 2, should read X = X + 2.

Armando L. Senson
Virginia Beach, VA 23464

FREELINE REVISITED

As a result of some recent thinking and experimenting, I would like to present the following changes to my Freeline (*SB* 5/95):

I added 37g (1.3 oz) of stuffing to the lower section of the line, raising the stuffing density to 1.125 lbs/ft^3 in that section. I also raised the upper section to 1.5 lbs/ft^3 by adding 25g (.9 oz). This has the effect of somewhat extending the bass response to a lower frequency (by means of slowing down the speed of sound in the line, which effectively "lengthens" it).

I was happy to be able to remove the 8Ω series resistor, and I also changed the value of the capacitor in the treble circuit from 3.3µF to 4.0µF.

Upon reviewing the article, I see that I failed to include a feature of the prototype Freeline. The original had an area of surgical cotton batting on two sides and the rear inner surfaces. This pad is about 8″ wide (or the width of the cotton role) and about ½″ thick. It should be centered on the woofer magnet, so that half of it is in the lower section of the line and half in the upper section. It can be made of one long piece, or it can be three shorter pieces. The latter might be more practical for a retro-fit of a completed Freeline. Fortunately, due to the configuration of the Freeline, it shouldn't be hard to insert. It could be spot-glued in position, or maybe the stuffing would be dense enough to hold it in place.

I'm so used to designing systems that go up against the wall behind the speaker that I forgot the Freeline has a different configuration than my others. I have found that the Freelines (especially the modified ones) sound better to me if the rear of the bases are a foot or more from the wall.

I think if you construct the Freelines, you will be pleased with these modifications. I found the results so impressive that I added four more ounces to the stuffing of my Squatline—the system I used in my experiments in designing the Simpline Sidewinder Woofer (*SB* 4/95) for reader Harry Campbell. This resulted in a stuffing density of about 1.375 lb/ft^3. Prior to this I tried stuffing it to a density of 1.5 lb/ft^3, but this was too much, making it sound a bit "stuffy." The four-oz addition, however, seemed right on.

I talked Harry into adding the extra stuffing to his Sidewinders, and he sounded very happy with the change. We both agreed that the 4Ω series resistors in the Sidewinder circuit were no longer needed (if indeed they ever were).

Fired up with these successes, I decided to play a bit with the Simplines. I noticed a decided improvement when I added 47g (say 1.7 oz) of stuffing.

I had previously made a retro-fitted improvement to the Simplines and submitted

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I certify that the statements made by me above are correct and complete.
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it as an article to Ed Dell. I think it is scheduled for publication in early 1996.

The above changes came about from offshoot experiments on a recent project which resulted in the development and construction of a new transmission line I call the B-line. The prototype is a woofer for a 6½″ speaker that goes lower than I have heard a small woofer go before. I am currently incorporating this prototype into a breadboarded system to obtain data for a folded full-range B-line. When I feel that things are ready, I'll probably present an article on these.

John Cockroft
Sunnyvale, CA 94087

IN PRAISE OF PVC

I am writing in response to John Cockroft's letter (*SB* 6/95) in which he offered suggestions as to why W. Werner's version of Cockroft's Simpline was not totally successful. (Werner noted that the bass was muddy.) Cockroft believes the schedule 40 PVC tubing used for Werner's enclosures is the cause of this problem. Cockroft wrote that this material is not suitable for loudspeaker enclosures because it vibrates with the speakers (presumably more than another enclosure material).

I disagree with this. I have built a pair of sealed enclosures that use 5″-diameter, schedule-40 PVC tubing and fittings. There are two 5¼″ woofers in each enclosure. Having listened to them for a long time, I can attest that the bass is tight, little enclosure vibration can be felt, and enclosure resonances are audible only when my ear is actually on the enclosure.

There are two primary reasons why I built an enclosure out of PVC tubing and fittings. First, it was fairly simple to build—really just a matter of cutting the pipes and fittings to the correct length and putting them together. (Few PVC-tubing enclosures in *Speaker Builder* took advantage of the fittings that are available.) Granted, there is some wood (specifically hardwood plywood) used in this enclosure. The drivers are bolted to plywood rings, which are in turn caulked to the tubing. Also, the enclosure base is made of plywood.

The second reason for using PVC tubing is that enclosure resonances of a tubular enclosure can be considerably less when compared with a typical box design. If a panel (side) of a box enclosure were studied by itself, you would find that it is essentially clamped around its edges but free to vibrate in the center. The panel is similar to a drumhead. Therefore, when pressure changes occur in the enclosure, the panel moves in and out, thereby leading to sound coloration. Even worse, the panel dimensions lead to coloration at specific frequencies. In an enclosure with a tubular cross-section, however, pressure changes essentially force the tube to expand and contract. This expansion and contraction action is much more difficult than the bending action that takes place in a "box" enclosure. This is one reason why pressure vessels have circular cross-sections.

For these reasons, I encourage *SB* readers to consider using PVC tubing for their next enclosure. The tubing is available in a wide variety of sizes, and I have been told that fittings (such as "tees," elbows, couplers, etc.) are available for all existing tube sizes. I hope to build one or two subwoofers using PVC tubing for the enclosures.

My other reason for writing is because of G.R. Koonce's article entitled "The Waveguide Path to Deep Bass," also in *SB* 6/95. In this article, Koonce expresses his preference for using "small" drivers (and, presumably, small enclosures) for providing deep bass. Another way to meet these criteria is to follow the suggestions of Tom Nousaine's article, also in *SB* 6/95, entitled "Stereo Bass: True or False?" I felt that one very important point in this article was the suggestion to place subwoofers in the room corners, which raises their efficiency considerably. This is a very valuable relationship.

Suppose you place a bandpass subwoofer enclosure in the listening-room corner. Assume an efficiency of 90dB SPL to match the main speakers, and that corner placement is going to provide 15dB of gain. (I suggest using The Listening Room software by Ralph Gonzalez to find what the specific gain will be for your listening room. This is also a very good program for determining the theoretical room response for your speakers.) Now, a key value for determining the box parameters for a bandpass enclosure is the desired system efficiency, which, if you take the room gain into account, would be 90 – 15 = 75dB. An important relationship in bandpass-enclosure design is that as the desired system efficiency decreases relative to the raw driver efficiency, the resulting bandwidth shifts lower in frequency. Therefore, if the room gain is accounted for in a bandpass enclosure design, a small driver may provide suitably deep bass.

I encourage *Speaker Builder* readers to experiment with this type of design to reduce subwoofer size and cost.

J. David Keener
Lititz, PA 17543-8860

The Waveguide Path To Deep Bass

from page 40

waveguides *do* work, and the test structure did integrate well with a direct radiating system producing the remainder of the audio band. The waveguide must be very stiff and fairly accurately built as to length ratio of the two pipe sections. It is doubtful whether a passive crossover is practical with these devices.

I discovered that the general frequency response of the waveguide rose with frequency over the lower portion of the passband, and that best performance occurred with equalization (–6dB/octave) over this portion of the band. Close coupling between the waveguide and a single driver on both sides of the cone was never achieved; therefore, the single-driver configurations tested did not fare as well as other configurations. The use of dual drivers, driven out of phase and close-coupled to the waveguide structure, worked well, producing a passband roughly as predicted by the design equations from the references.

The introduction of a rubber throat between the waveguide and the dual drivers appears to somewhat reduce sensitivity to errors in the ratio of the two pipe lengths, with minimum loss of gain. Questions still exist as to the size of the waveguide pipe area, exactly how to implement the waveguide with a single driver, what usable sensitivity improvement (gain) is available, and what the displacement function is for drivers used in such structures.

I'd like to hear from anyone else experimenting with low-frequency waveguide structures.

Screws and Fasteners

from page 47

must decide all this before designing the speaker system, and make appropriate increases in the side, top, and bottom dimensions. It is a good idea to use washers to more evenly distribute the compression of the bolt, and a firm rubber gasket is most important. The spacing, S, of the hanger bolts should be about five times the thickness, T, of the back material (S = 5T).

My favorite attachment method for this type of enclosure is to use solid-brass inserts and bolts (*Photo 9*). These hollow cylinders are threaded on the inner surface to accept standard bolt and screw sizes, and have screw threads on the outer surface for gripping the wood. To prepare their insertion, set a stop on a brad-point drill to about 1/32″ longer than the length of the insert (*Photo 10*).

After drilling a perpendicular hole into the backstop frame, drive in the insert with as large a screwdriver as fits the slot (*Photo 11*). Normally, this is too large to completely drive the insert to just below the surface (*Photo 12*). If you must use a smaller screwdriver, be very careful, as the brass is quite soft. It is usually easier to put the insert into the backstop frame before gluing the frame inside the enclosure, as indicated in *Photos 11* and *12*. *Photo 13* shows the relationship of the hanger bolts or insert anchor bolts (screws) to the speaker enclosure back, the enclosure, and the backstop frame.

To summarize: if you must use screws, remember to

- drill pilot holes of proper size;
- use high-RPM and slow feed rates when drilling;
- use hanger bolts or screws with inserts for repeated-use applications.

C-101 Equalizer

from page 52

machinery, which measures the resistance (or capacitance) of the components before insertion to ensure high performance.

To answer the questions on how we (and most of the industry) measure signal to noise, the short answer is A-weighted. This weighting mimics the way we hear at normal listening levels and takes into account the Fletcher-Munson effect. To double-check our published specifications, we measured a number of units from our normal production with our recently tuned-up Audio Precision System One. The result for the 2V that Gary measured was –109dB signal to noise. We should mention that these measurements are made with the spectrum display off, which, we believe, is the way a critical listener would use the product. With the display on at 2V input, our measurement is –104dB and within our published specifications.

Finally, we strongly agree with Gary on the subject of making room acoustics as good as possible. An equalizer is one of the tools of good sound (as shown in the Canadian National Research Council Athena project), but, as we mention in the manual, the first rule of good equalization is to work with speaker position and room acoustics. However, we disagree with Gary about limiting the C-101 to receiver-based systems. Based upon their comments on many, many warranty cards, our customers are quite satisfied. We will be happy to provide a copy of those warranty-card comments to anyone who contacts us.

Tom Walker
President
AudioControl

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Welborne Labs

Sony tape reel-to-reel model TC 263D matching stereo recording amp SRA-2 (tube) with all manuals, excellent condition, $175; 15 master tapes including Polk C4 audio bandpass sub box, 200W, fits trucks nicely, $250 with warranty card & original box. Cliff Steele, 865 Donelda Lane, Rockford, IL 61102, (815) 965-9180.

Bose 901 series VI with EQ, brown, all unused and still in their boxes, $800 and you pay shipping. Andy, (360) 321-7852.

Alpine amp 30WPCX2, $100; two Autotek car amps 60WPCX4 or 150WPCX2 two-way crossover built-in, $325 each ($580 new); 3-4-5 way crossover, $120 (paid $200); JVC KDGS550 in-dash CD, never installed, $240; two pair Aurasound Bass Shakers, $90/pair. All items mint. Phillip, (615) 791-6446 after 6 p.m. CST.

AIRR (Anechoic and In-Room Response) software, $30; Dynaudio D54, $70 each; MediaVision Pro-Graphics 1024 Vesa video card with 2.25M-byte RAM, $75. Dan Patten, 1768 N. 980 W., Orem, UT 84057, work (801) 224-8080 or home (801) 225-8577.

Four Dynaudio Esotec D260, $50 each; two Dynaudio D52AF, $55 each. Brad, (513) 433-6229.

Two each Dyna ST-70 power amps, PAS-2 preamps, and FM-2 tuners, extra tubes, manuals, good condition, all $265; NEC Dolby Pro-Logic surround decoder, full remote, excellent condition, $70; two 16″ x 36″ tube subwoofers, 10″ Pyle drivers, crossover, 96dB SPL, good solid bass, $100. Buzz, (301) 872-5982.

Five pair Stratheam ribbon drivers in good condition, $250 per pair plus shipping. Andy, (414) 466-8266, eves. (Wauwatosa, WI).

Pair VMPS larger subwoofers, walnut, $350 each plus shipping; Tandy 3900 laptop, 4M-byte 120HD, Windows 3.1, new batteries, warranty expires 12/96, with case, $800; Kenwood car amp KAC644, 40W x four channels or 80W x two, $125; Kenwood KDSP100 surround processor with EQ, $125. Steve, (219) 531-0221.

Tannoy dual concentric loudspeaker with crossover unit, types LSV/H8 15/8, great shape, $340; Carver magnetic field power amp M-15t, 1,200W RMS, 350W/channel, $450. Matt Edwards, 329 E. Hampton, Stockton, CA 95204, (209) 463-4926, leave message.

LMS card, includes software, manual, and cables, comes installed in a Compaq portable 286 computer, $800 or best offer. Joseph, (407) 575-5819.

Pair Klipsch Klipschorn copy, black, Stevens woofers, Altec 802 & 511 horns, custom crossover, $599 (pick-up only); Conrad-Johnson Motif MC-9 preamp, $495; Altec Lansing 1653A EQ, $195; pair NAD Series-20 speakers, $75 (pick-up only); Crown VFX2A crossover, $149; Denon DCD-1560 CD, $250. Scott, phone/FAX (310) 835-2967.

Altec 421-H 15″ woofer, $75; 902-8T driver, $65; pair 511B horns, $75; Soundolier FC-105 heavy 5″ fullrange speakers, new, $12.50 each; pair Stanford Acoustics 15″ cast frame woofers, 3″ voice coils, as new ($450), $160; four Bozak 12″ woofers and 12″ tweeters, $225. Jim, (360) 466-5161.

Sell or trade: Compaq 286-12 Deskpro with math co-processor, 2M-byte memory, EGA, Q-40 tape driver, HD floppy, slow but reliable, runs Windows 3.X, $300 or trade for old Dynaco or equivalent equipment. Rob Lewis, 2950 N. Dobson #11, Chandler, AZ 85224, (602) 963-8833, FAX (602) 963-3766.

Pair Cabasse 21NDC 8″ woofers, $250; Parasound 1200II amp, 15 months old, $650 or best offer; B&K MC101 Sonata preamp, five years old, $450 or best offer. Don Vogel, 1461A Fifth Ave., Ft. Knox, KY 40121, (502) 942-5039.

LMS System V3.5, $800 or with stocked 486/66, $2,200; Boston Acoustics Lynnfield 500L Series II, rosewood, perfect, lists for $5,500, selling for $2,750; Sony WMD-DT1 portable DAT player, list for $500, selling for $260. Stephen, (415) 721-0553 or mrmugwort@aol.com.

LMS system, complete including mike, $600. Gordon, (408) 946-5993.

Dynaudio D21AF: brand new, in carton, $35/each or used, $20/each; LPG Titanium dome tweeters 26T, $20 each (excludes shipping). Fred, (610) 693-6167.

Pair Scan-Speak 18W8546 Kevlar 7″ woofers, new, $200. Edward, (206) 562-9848.

Four Pioneer 75W/channel stereo power amps with variable damping (for you tube freaks) and variable input sensitivity, $200 each or $700 all (instant surround sound); pairs of Altec 802, 808-B, 806 horn drivers; 811 horns; N-809-8A crossovers; Voice of Theater speakers; JBL horns, woofers. David, (914) 688-5024.

Mint pair Dynaudio 30W100 woofers, never used, in original boxes, $150; pair Technics Leaf tweeters, mint, $75; Sonic Frontiers SFL-1 Signature tube preamp, new, in original box with all documentation, $1,000. All prices or best offer. Bill, work (702) 385-7170 or home (702) 647-2563.

WANTED

Altec 1566, 1567 mixer; 436, 438 compressor amp; RCA MI9449 15's; UREI, White, Pultec, dbx, Langevin, Fairchild EQ or compressor. Kent Elliott, phone/FAX (913) 677-1824 (KS).

Western Electric 713-C compression driver. E. Billeci, (503) 234-8453.

Ad Index

Tube hi-fi, theater amps/preamps and speakers: Altec, Jensen, McIntosh, Marantz, Craftsman, Dynaco, Heath, Eico, etc. ***SUNSHINE SOUNDS***, (405) 737-3312.

Tektronix 475(?) transistor/tube curve tracer; tube amps with potted output transformers (amps can be dead but transformers must be OK); NOS potted output transformers; oil caps. Robert, phone/FAX (707) 523-2921.

Celestion DL4 Series II loudspeakers; Magnavox CDV488 combi player. Robert Irwin, 936 Pearson Rd., Paradise, CA 95969, (916) 877-7734.

Information on the Curcio version of the Swan IV Satellites, need wiring information and data on the crossover used for this version. R. D. Darroch, 1807 Elm Crest Dr., Arlington, TX 76012-1908.

Crown OC 150 control center, must be in good condition; Dahlquist DQ MX1 passive matrixing crossover wiring diagram or information. Bill Schulte, 3566 Pape Ave., Cincinnati, OH 45208, (513) 321-3175.

Schematics and any service information for G.A.S. power amps (Ampzilla or Son of Ampzilla), parts cross references would also be handy; Realistic Mach One speakers, catalog #40-4024A only, blown tweets or pads OK. Rick, (507) 625-9372 after 5 CST or ark-grafix@aol.com.

Altec 291-16A drivers; Altec 291-16A diaphragms (stamped #21531); JBL 2402 "Bullet" tweeters and diaphragms; Crown Macro-Tech 600, 1200, 2400 amps, blown or scratch & dent okay. Bob, (804) 523-0711.

Info on Onken speaker enclosures; specs on EV EVM 15L; single-ended triode amp schematics. Dan Hon, #12619-074, PO Box 2000, Millington, TN 38053-2000.

JBL drivers, dead or alive: 2123H 10" mids, 2450J horn drivers, 2382A horns, 2220J 245; Altec 808 kits or replacement; frequency generator and counter with .1Hz resolution(audio); used LMS or woofer tester, IMP or similar system; RCL meter; motor for Connoisseur BD1 turntable. Don, (905) 824-9980, CIS 74260,1115.

Orion Audio Blue Book, Knight KN 612 HC coaxial speaker; Calrad CH-300 multicellular horn; Altec 4024A signal processor; diaphragms for Altec compression drives. Bob, (916) 273-9679.

Plans for the construction of Klipsch Bell Klipsch speaker; working SpeakerLab midrange drivers. Don Williams, day (804) 447-7100 or night (804) 738-6510.

Any info, suggestions, or experience with Stephens 80FR fullrange speaker (16Ω type). I need suggestions on proper enclosure size, venting, porting, etc. I'm willing to pay for copies of any data from Stephens you may have on this or other Stevens FR series. Albert Mitchell, (212) 865-4478, FAX (212) 662-1823.

CLUBS

If you are interested in joining a club or forming one in your area, contact the following:

ARIZONA

Arizona Audiophile Society
PO Box 13058, Scottsdale, AZ 85267
Paul Christos, (602) 971-3979

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65 Washington St. #137, Santa Clara, CA 95050
John Atwood, FAX (408) 985-2006

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Dave Clark, (310) 427-4207

San Diego Audio Artisans
Dennis, (619) 297-3352 eves.
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Chris Clarke, FAX (011) 27-21-7618862,
E-mail chrisc@iaccess.8a

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