BLASTER TESTING FOR SYSTEM QUALITY

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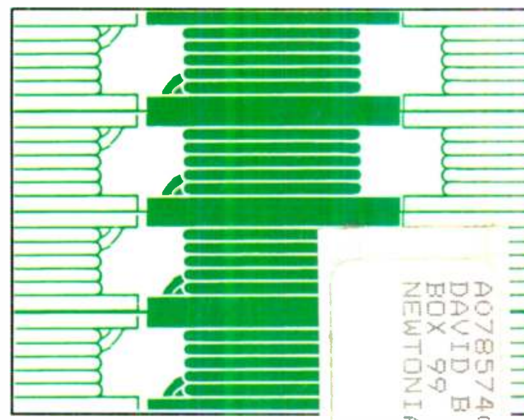
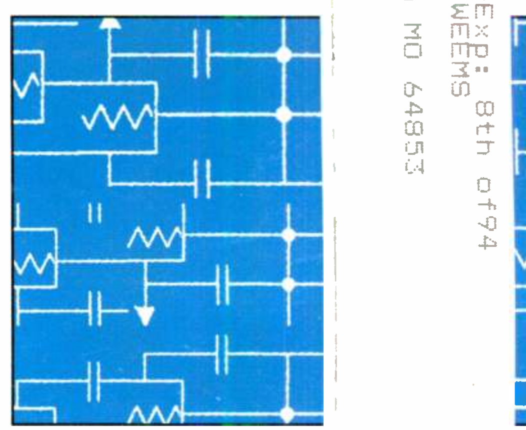
Speaker Builder

THE LOUDSPEAKER JOURNAL

$75 RESURRECTION FOR JUNKED AR2ax PAIRS



THE LINEAR ARRAY BECOMES HARDWARE



LOUDSPEAKER 101: PIERCE ON THE Qs

CHECKING YOUR SYSTEM WITH *PC AUDIOLAB*

MAKING A FRAME FOR THE GRILLE

ARE HORNS GOOD STUDIO MONITORS?

Good News

IMPROVED SURROUNDINGS

Canadian PSB Speakers has debuted two new models designed for use in the center channel of Dolby Surround Pro Logic and surround-sound home theater systems. The PSB 200C is a vented design with dual 5.25″ injection-molded polypropylene drivers and a centrally mounted ½″ poly-flare tweeter. The Stratus C5 has dual horizontally deployed 5.25″ mid/woofers and two central, ferrofluid-cooled, ½″ poly-flare tweeters. PSB International, Inc., 633 Granite Ct., Pickering, ON L1W 3K1, Canada.

Reader Service #101

RACK ATTACK

Sescom Electronics' new product brochure features the Rackem 'N' Stackem™ series of half-rack audio accessories. Designed to work in a very compact environment, the units can be placed on a tabletop, in small tabletop half-racks, full standard racks, or wherever convenient. Other products listed in the brochure include switch boxes, stereo crossovers, dual-mike line drivers, six-channel headphone stereo amplifiers, mono monitor bridge adapters, relay boxes, stereo phase meters and tone controls, and audio testers. Sescom, Inc., 2100 Ward Dr., Henderson, NV 89015, (800) 634-3457, FAX (800) 551-2749.

Reader Service #116

DEVELOPING AIR MASS

Beals Brothers Acoustic Design has developed Air Mass Technology for auto sound products. With AMT the air mass acts as an acoustic amplifier, allowing the system to generate a lot of bass with not much driver cone motion. The results are lower distortion (second harmonic lowered by 20dB), greater output (3–5dB), and greatly reduced enclosure size. Beals Brothers, 2165 Lehigh Station Rd., Pittsford, NY 14534-2675, (716) 367-2610, FAX (716) 334-8215.

Reader Service #108

SAW SUBSTITUTE

The Spira Cut System from Roto Zip features a complete line of Zip Bits for fast, precise cutting through ceramic tile, paneling, plastic laminates, plywood, drywall, and other materials. The hand-held, high-speed tool does the work of a jigsaw, reciprocating saw, router, and sabre saw. Roto Zip Tool Corp., 1861 Ludden Dr., Cross Plains, WI 53528, (800) 521-1817, (608) 798-3737, FAX (608) 798-3739.

Reader Service #112

FRIED CONES

The new drivers available from Fried Products, including those incorporating the m.a.r.s. technology, offer: a cone material free of high-frequency peak; a magnetic system which is linear at all excursions; and a four-layered voice coil system with a driver Q of approximately 0.2. Fried Products Corp., 1323 Conshohocken Rd., Norristown, PA 19401-2707, (800) 255-1014, FAX (610) 277-4390.

Reader Service #114

Speaker Builder (US ISSN 0199-7920) is published every six weeks (eight times a year), at $32 per year, $58 for two years, Canada add $8 per year; overseas rates $50 one year, $90 two years, by Audio Amateur Publications, Inc., Edward T. Dell, Jr., President, at 305 Union Street, PO Box 494, Peterborough, NH 03458-0494. Second-class postage paid at Peterborough, NH and an additional mailing office. POSTMASTER: Send address change to: *Speaker Builder*, PO Box 494, Peterborough, NH 03458-0494

← Reader Service #36

Good News

■ BASS BOOSTER

AudioControl's latest addition to its Performance Advantage line is the Phase Coupled Activator Series Three. The digital subharmonic restoration system has speaker level inputs that allow you to connect the unit and a subwoofer to any system, even one without preamp outputs for a subwoofer. Other features include a 24dB/octave Linkwitz-Riley electronic crossover with frequency programmability. AudioControl, 22410 70th Ave. West, Mountlake Terrace, WA 98043, (206) 775-8461, FAX (206) 778-3166.

Reader Service #102

■ NEW GALAXY

Galaxy Audio has acquired Valley Audio Products of Merriam, KS, and will be relocating the business in Wichita. Under its new ownership, the company will be known as Valley Audio. With a product line dating from the 1970s (it was originally known as Allison Research), Valley Audio introduced the first dynamics processing capabilities for studio recording. The company manufactures electronic signal processors including the Kepex®, Gain Brain®, Model 401 microphone processor, Dyna-Mite®, X-Gate, and the Model 730 Dyna-Map™ digital processor. Valley Audio Products, 625 E. Pawnee, Wichita, KS 67211, (800) 800-4345, (316) 265-9500, FAX (316) 263-0642.

Reader Service #113

Continued on page 8

➲ REMOTE CONTROLLED

Parasound has introduced its first multimedia processor/preamplifier, the P/SP-1000. The Dolby™ Pro Logic A/V control center offers complete remote operation of customized home theaters. It comes equipped with three audio/video inputs, with switching capability for both NTSC composite and S-Video, and two audio inputs which are compatible with any typical analog line-level source. A "preamp direct" switch controls a straight-wire connection from the source to an exterior amp. Parasound Products, Inc., 950 Battery St., San Francisco, CA 94111, (800) 822-8802, (415) 397-7100, FAX (415) 397-0144.

Reader Service #103

■ CABLE NEWS

GOERTZ MI loudspeaker cables employ the "Laminax" technology of combining metal foil and strip conductors with a variety of dielectric materials (used in electronics for EMI and RF shielding purposes and as winding conductors in high- and line-frequency transformers and inductors). The result is a thin cable with high dielectric strength. The conductors have a low characteristic impedance that matches that of loudspeakers, which the makers claim results in virtual elimination of distortion. Alpha-Core, Inc., 915 Pembroke St., Bridgeport, CT 06608, (203) 335-6805, FAX (203) 384-8120.

Reader Service #106

■ FRESH G.R.A.S.

Scantek has released the latest series of microphone preamplifiers manufactured by G.R.A.S. Sound & Vibration: Types 26AA, 26AB, and 26AC. The units are ideal for 1″, ½″, ¼″, and 1/8″ microphones, with or without polarization voltage, and for high-impedance input stages for instrumentation. They can also be used for accelerometers and hydrophones. Their frequency range is from less than 1Hz to greater than 200kHz, with supply voltages from below 28V to above 130V.

Reader Service #109

Another recent G.R.A.S. release from Scantek is the Type 50AI sound intensity probe. Based on a new microphone venting system, the unit has built-in remote control for the Norsonic 840, 01dB, and other leading intensity analyzers. The 50AI comes equipped with a pair of Type 40AI Sound Intensity Microphones and stainless-steel Type 26AA microphone preamplifiers. Scantek, Inc., 916 Gist Ave., Silver Spring, MD 20910, (301) 495-7738, FAX (301) 495-7739.

Reader Service #110

About This Issue

Fed up with all the blather about Chuck and Di and Burt and Loni? Well, some marriages are more compatible. Witness computers and audio. The union of these two technologies to perform speaker measurements is well-represented in this issue.

Julian Bunn introduces his high-flying AIRR ("Anechoic and In-Room Response," p. 10) measurement program, which takes advantage of the capabilities of the popular SoundBlaster sound card on your PC. With this program, you can turn your computer into an inexpensive tool for surprisingly sophisticated system analysis.

Another example of our reliance on computers in audio work is PC AudioLab from Microacoustics. **Richard Campbell** reviews this program ("Software Report," p. 49), which offers PC readers a handy, low-cost solution for system measurements.

In this age of environmental awareness and recycling efforts, it's appropriate to show you how to inject new life into old speakers. **Hilary Paprocki** ("Reviving the Classic AR2ax," p. 20) guides you through the restoration of Acoustic Research speakers in 15 easy-to-follow steps.

SB is pleased to reprint, from the March 1994 issue of *Studio Sound*, the results of an exhaustive five-year study of horn loudspeakers to determine their suitability for studio use ("Round the Horn," p. 24). The results of this research, which one of the authors used as the basis of his doctoral dissertation, are significant and should affect future designs. **Philip Newell** and **Keith Holland** also detail the horn cutoff phenomenon, a little-understood, but important, key in studio design.

Philip Witham returns...this time with a prototype. And while he plans more testing of his 60-channel dipole speaker, initial results indicate that his Linear-Array concept works to produce a better, "more realistic, more pleasing" sound than with stereo. The next steps involve further tweaking, building, and electronics. Stay tuned.

Dick Pierce has a lot on his mind these days. He takes the opportunity in this issue to explore such diverse topics as free online advice (sometimes worth every penny) ("Guest Editorial," p. 9) and the Q-rious significance of Thiele/Small parameters ("Loudspeakers 101," p. 70).

In his latest installment ("Wayland's Wood World," p. 47) woodsmith **Bob Wayland** offers some useful tips for putting the finishing touches on speaker projects.

Speaker Builder

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

Speaker Builder is published eight times a year in the interest of high-quality audio reproduction.

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Speaker Builder

THE LOUDSPEAKER JOURNAL

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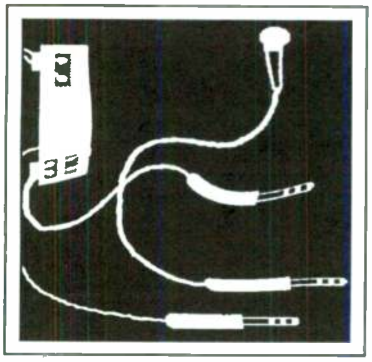

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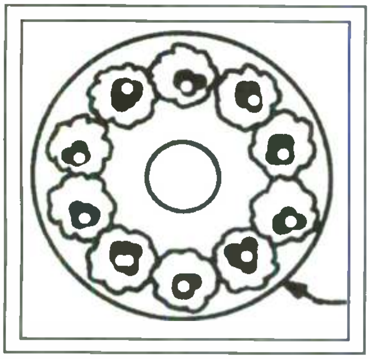

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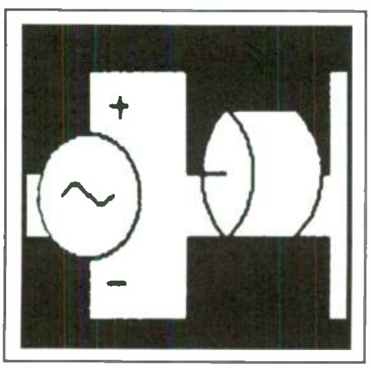

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DEPARTMENTS

Good News

■ ER OPERATIONS

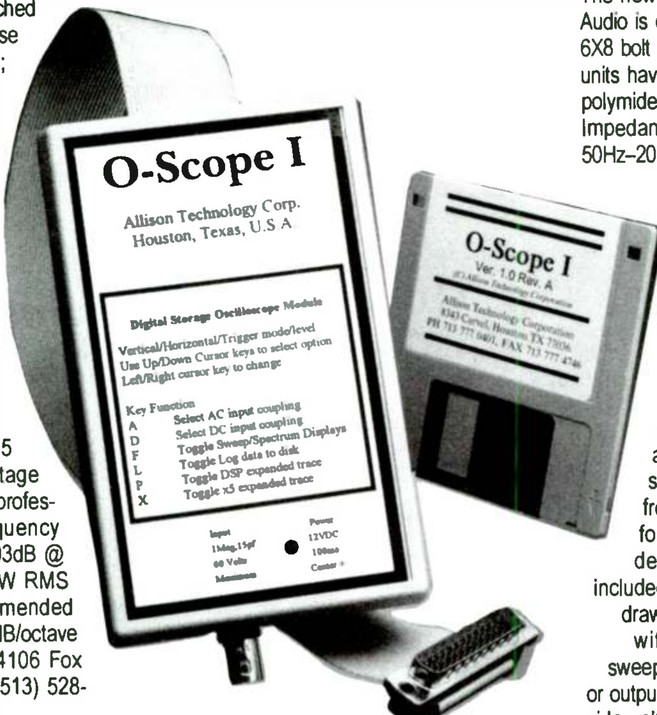
Recent technological advances in electrorheological (ER) fluids offer performance features such as operation over a temperature range of -50°C to 150°C; finer-particle formulations, carefully matched densities, and other techniques to decrease settling problems and increase stability; softer, nonabrasive particles; and higher shear stresses. ER-based adhesives and coatings, when used on nonhorizontal surfaces, convert to a higher viscosity when an electrical field is applied. Call Technology Catalysts International at (703) 237-9600, FAX (703) 237-7967, for details.

Reader Service #111

■ COMPACT DRIVER

The newest design in high-frequency ribbon loudspeakers is the SA 8535 Neodymium Compact Driver from Stage Accompany, a Dutch manufacturer of professional sound systems. It has a frequency range of 1kHz–30kHz, sensitivity of 103dB @ 1W/1M, and a power handling of 60W RMS with 1kW peak (200ms). The recommended crossover point is 1kHz or higher (12dB/octave minimum). Stage Accompany USA, 4106 Fox Run Trail #6, Cincinnati, OH 45255, (513) 528-4035, FAX (513) 528-4037.

Reader Service #104



■ INSTALLATION BREEZE

The new 9757 two-way speaker from Fultron Car Audio is designed to match virtually every 5X7 and 6X8 bolt pattern on the market. The easy mounting units have polypropylene cones, flexible surrounds, polymide dome tweeters, and strontium magnets. Impedance is 4Ω; resonant frequency of 75Hz; 50Hz–20kHz frequency response, with a maximum power handling capability of 80W. Sensitivity is 88dB. Fultron, 122 Gayoso, PO Box 177, Memphis, TN 38101-9988, (901) 525-5711, FAX (901) 525-7993.

Reader Service #105

⊂ SCOPING PCS

The O-Scope I converts your PC into a digital storage oscilloscope. It captures and displays DC, audio, and low-end ultrasonic frequency input signals. A 128-point frequency spectrum analyzer mode, suitable for use in measuring distortion to 1% and determining frequency response, is also included. The device is powered from 12V DC, draws less than 40mA of current, and works with AT-compatible computers. Trace sweeps can be frozen onscreen, saved to disk, or output to a printer. The O-Scope I can also provide voltage, frequency, and period calculations. Allison Technology Corp., 8343 Carvel, Houston, TX 77036, (800) 980-9806, FAX (713) 777-4746.

Reader Service #107

Guest Editorial

The Newbies Are Back

By Richard Pierce

Well, it's the fall semester, and all the pubescent newbie experts are back in town, a-rarin' to go. They bring with them the same old notions the last ten generations brought. They're clueless, and the thing they are most clueless about is exactly how clueless they really are.

I have just shut down two private e-mail threads from two kids whose combined age is less than my own, and neither of whom was even born when I was doing loudspeaker design consulting.

For example, one of these, a 19-year-old college sophomore who admits to no knowledge of physics or electronics, who conceded to not knowing the technical lexicon of loudspeakers and acoustics, whose *only* reference text on loudspeakers is Vance Dickason's *Loudspeaker Design Cookbook*, and whose only qualification seems to be as a "$45 per hour car audio installer," chose to try to re-educate me on how speakers work. He was sorry I was so stupid, but all I had to do was to just look up in Dickason what he was talking about. On at least two occasions I tried to get a specific page citation, but he seemed strangely silent, then proceeded to redefine what "isobarik" meant in the context of loudspeakers.

It's something of a small tragedy when this happens, like clockwork, every September, when a new incoming class discovers, simultaneously, the Internet, their interest in some topic, and testosterone, and confuses all. I clearly remember when I was 19, and I knew *absolutely everything,* and I was so soundly trounced in a technical discussion, there was almost no body left to identify.

The first thing I definitely knew after that was that I definitely *did not* know *absolutely everything,* and that piece of knowledge has stayed with me as I strove in the subsequent years to try to learn absolutely everything.

WRONG IS WRONG

Yes, some of you may say, this is all true, but aren't we being a bit hard on these kids? After all, what harm can they do?

Well, in these cases, they were offering advice to people with specific questions, and their answers were just plain wrong. Wrong is wrong, whether it comes from the mouths of babes or adults. Wrong is wrong, because the physical reality of loudspeakers, and all other physical realities, pays no heed to either good intentions or a child's innocence.

And, when someone else with good intentions takes that wrong answer and then tries to make it work, and it doesn't, the outcome is, as often as not, discouragement resulting in that person assuming, quite incorrectly, that *he* is at fault, not the person who provided the wrong information, and he is thus deterred from pursuing what may be a good path.

The person cannot be excused simply because he didn't know better. Wrong is wrong, no matter what the source. And wrong information has no business in a technical news group, no matter how good the source's intentions.

CHEAP EXPERTISE

Vern Mastel, in a recent guest editorial in *TAA* 2/94 ("Hoodwink City USA," p. 7), was most articulate about the subject:

"...Cheap expertise...It is not something to be avoided, it is to be attacked like a cancer. Far too many armchair experts are spouting opinions, although their expertise is often limited to what they have read or heard. True experts hold their opinions very dear and don't broadcast them unless they are accompanied by substantial experience and data. 'Cheap expertise' is always offered on the basis of 'knowing it all.' The true experts have an appreciation for how much they don't know."

Another area Mastel deals with is the subject of one of these people pressing on regardless of his lack of familiarity with the terminology:

"Avoid technical shorthand? No, up to a point. The standard practice of the technologically illiterate is to overcomplicate the simple and oversimplify the complex. Correct terminology cannot be compromised. Converting technical concepts into 'layman' terms often results in technical nonsense [as it did in the case at hand—rdp]. The intent of correct terminology is not to bury the novice, but rather to properly express a concept or discuss a design."

Yes, Vance Dickason's book is a good book—probably the best of its kind—for its purposes, which is as the title says, a "cookbook" intended specifically and exclusively for amateur loudspeaker enthusiasts. But it is hardly the reference text on loudspeaker theory.

So, our young friends, welcome to the fray, and join us who have been here for a long time. But watch where you step, because you may learn firsthand where the term "hooey" comes from.

And, I suppose, for the rest of us, some patience.

AIRR: ANECHOIC AND IN-ROOM RESPONSE MEASUREMENT

By Julian J. Bunn

After enjoying Bill Waslo's articles on the IMP system ("The IMP," *SB* 1/93, 2/93, 3/93, 4/93, 6/93), I realized that the SoundBlaster™ card I recently installed in my PC might be able to perform similar analysis functions, under software control.

SYSTEM FEATURES

I drew up some specifications for a suitable program to include:

1. Facilities for measuring both loudspeaker and amplifier response as a function of frequency;
2. The possibility of subtracting the two;
3. An oscilloscope-like display of the audio signals being picked up by the card;
4. A graphical display of the frequency response;
5. Some means of adjusting the range of frequencies that would be covered.

To restrict the portion of the digitized signal to exclude the part caused by reverberation in the listening room, the tool needed to allow the signal to be "windowed" by adjusting graphics markers on the display. I was interested in measuring response to pulse signals and also to signals from a test CD, for example, sine tones, pink and white noise, and so on. Whereas the pulse analysis would require averaging in the time domain to reject background, these other signals would require averaging in the frequency domain. An important capability was to be able to switch between the two.

Finally, I required that the lowest resolvable frequency should be adjustable, maybe at the expense of data acquisition speed. The result was the AIRR (Anechoic and In-Room Response) measurement program.

ABOUT THE AUTHOR

Julian Bunn was born in Yorkshire in 1959. He studied physics at the University of Manchester and subsequently earned a Ph.D. at the University of Sheffield. He now works at an international organization in Geneva, and lives with his wife and two daughters in a small village in nearby France.

REQUIREMENTS & OPERATION

Despite not being as complete or professional a tool as the IMP, AIRR will allow you to make quite sophisticated audio response measurements. It may therefore be attractive to some readers as a "poor man's IMP." (I resisted the temptation of naming the program PIMP!)

To use the program, you will need a DOS-based PC, a SoundBlaster (or compatible) sound card, EGA or VGA graphics, a microphone (usually supplied with the sound card), and about 500K of free disk space. A math co-processor is *not* required. The faster your machine, the better the results from AIRR will be.

Readers of the IMP articles in *SB* will be familiar with the basic principle employed by AIRR, which is to produce a single pulse of very small width and then feed it to the audio system to be measured. The sound reproduced by the audio system is then picked up, using either a microphone or a direct electrical connection, and digitized by the sound card. The data is passed to a fast Fourier transform, and the frequency decomposition of the sound is obtained. This is then plotted in a graphics window on the PC, using the traditional logarithmic axes.

SOUND CARDS

A sound card is a printed circuit board in a slot on the motherboard of a personal computer and allows both recording and playback of digital sound. The card often comes as a package of parts including the electronics, a microphone, connecting leads, and a bundle of software. The board itself sports a sophisticated lineup containing an analog-to-digital converter (ADC), digital-to-analog converter (DAC), frequency modulation (FM), and mixer chips.

These days, the digital components are predominantly available in 16-bit versions (thus, a "16-bit card"), although 8-bit cards are still available and may be picked up very cheaply (but will not work with AIRR).

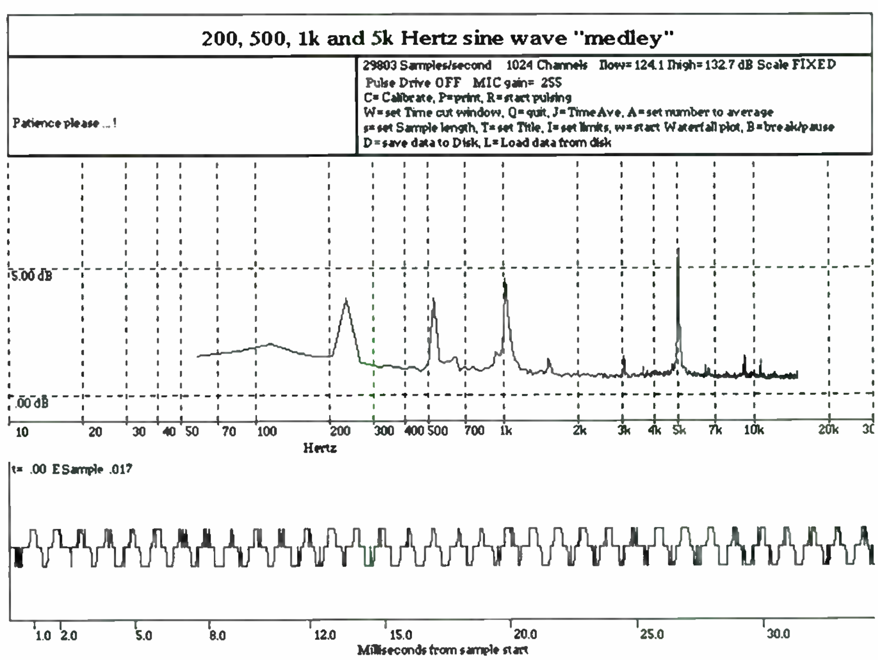


FIGURE 1: AIRR results from frequency averaging signals picked up via the microphone for four sine wave tracks played from a test CD.

Typical use of a sound card is for sound "bite" playback and multimedia applications such as games, voice recognition, document annotation by voice, interfacing to a CD drive in the PC, and so on.

The sound card may also be interfaced to and controlled by custom-built software. Sounds picked up by the microphone, after on-board digitization by the ADC, can be processed in just about any way that we wish. By suitably programming the on-board mixer, the signals from the line-in socket on the card can be treated instead of those from the microphone, or the two inputs can be mixed together.

Conversely, by programming the DAC, you can synthesize and feed to the line-out socket any desired waveform. (There is, however, some upper limit on the frequency of the signals that may be produced; I will cover this point later.) So, for a modest outlay of $100 or so we have a piece of equipment to perform arbitrarily complex digital signal processing.

PROGRAMMER WOES

Unfortunately, it's not quite as straightforward as that! Sound cards are notoriously difficult to configure correctly, and it is rumored that over 50% of them are returned "broken" to the vendor because of configuration errors. Why?

The answer is that the software drivers that come with the card, and which take care of how data is transferred to and from the electronics, require you to correctly specify several pieces of configuration information before the card will function properly. You, as the installer, are presented with the often confusing task of correctly setting Interrupt Request Lines (IRQs), Direct Memory Addresses (DMAs), and I/O address possibilities offered by the configuration utilities. In particularly unpleasant cases, you even need to change jumper settings on the board itself, if they conflict with other equipment settings. *[I strongly recommend Jeffrey Prosise's excellent volume titled "DOS 6 Memory Management" (ZD Press, Emeryville, CA, $29.95, ISBN1-56Z76-097-1), which explains not only memory management in DOS 5 and 6 but also clarifies matters of DMA and IRQs thoroughly and clearly.—ETD]*

Even with a correctly installed card, you may encounter problems such as insufficient memory for running CD-based multimedia sound applications, "choppy" audio due to badly sized buffers, and so on. (Having said all this, you may be lucky and discover that the card slots in and works the first time. Of the four cards I have installed, this happened just once.)

Once you have dealt with these problems, the next major hurdle to overcome is obtaining the necessary information on programming the card. It is rare indeed to find this sort of detail in the card's documentation. Obtaining it usually involves a visit to the relevant forum on CompuServe™, or downloading the appropriate FAQ (Frequently Asked Questions) list on the Internet News groups.

Finally, with a correctly configured sound card, the programming information described, and a decent language compiler for the PC, you are ready to code up a custom sound application.

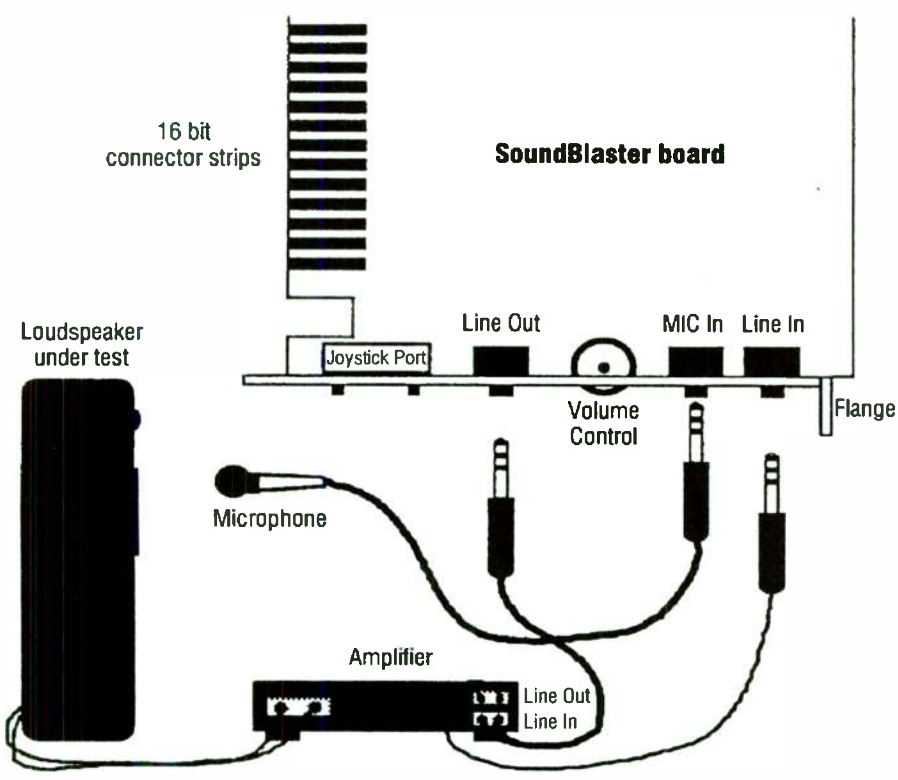


FIGURE 2: Showing the scheme for connecting external components to the sound card. Note that, to minimize distortion, the volume control should be positioned about midway.

CALIBRATING THE CARD

For this application, the most important aspect of the program to address is calibration. Unless only very subjective results are required, we must accurately measure how fast the card can digitize a sound, so we can calculate the sample rate. The sample rate achieved tells us the maximum frequency that may be resolved by the system (the lowest frequency is determined by the length of time we choose to gather samples).

Nyquist (of telephony theory fame) showed that for a maximum sampling frequency of W the signal bandwidth is W/2. For example, if the ADC on the card, under program control, can digitize and move a voltage level into main memory 20,000 times a second, then the maximum resolvable frequency is 10kHz. By calibrating the sample rate, the labels on the axes of the frequency and time plots can be positioned correctly, and we can be confident that a peak in the frequency response is really at the indicated frequency.

The AIRR calibration method uses the callable interface to the system clock on the PC to measure the time required to read into main memory a large number of 8-bit samples in so-called "direct-mode" ADC (as opposed to DMA mode). Using a large number of samples is important for two reasons:

1. The interface to the system clock only returns a system time measured to the nearest hundredth of a second.

2. The more measurements made, the smaller the derived calibration error.

The card is thus calibrated by (a) taking a reading, T1, from the system clock; (b) digitizing and moving 200,000 voltage levels in the ADC into main memory; and (c) taking a final reading, T2, from the system clock. The rate is then 200,000/(T2-T1). By default, the digitizations are made in blocks of 512 at a time. One block is henceforth referred to as a "sample" in the text, although it actually contains N separate measurements.

Several factors affect the number obtained for the rate:

- the speed of the software drivers controlling the card
- the speed of the sound card electronics
- the speed of the CPU and memory
- the efficiency of the machine code (the compiled program) that runs in the CPU.

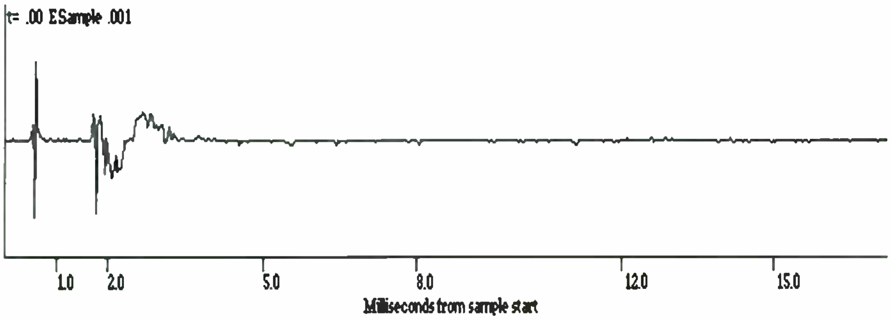


FIGURE 3: The result of time averaging pulses from the microphone. Note the crosstalk "blip" at around T=0.6ms (see text).

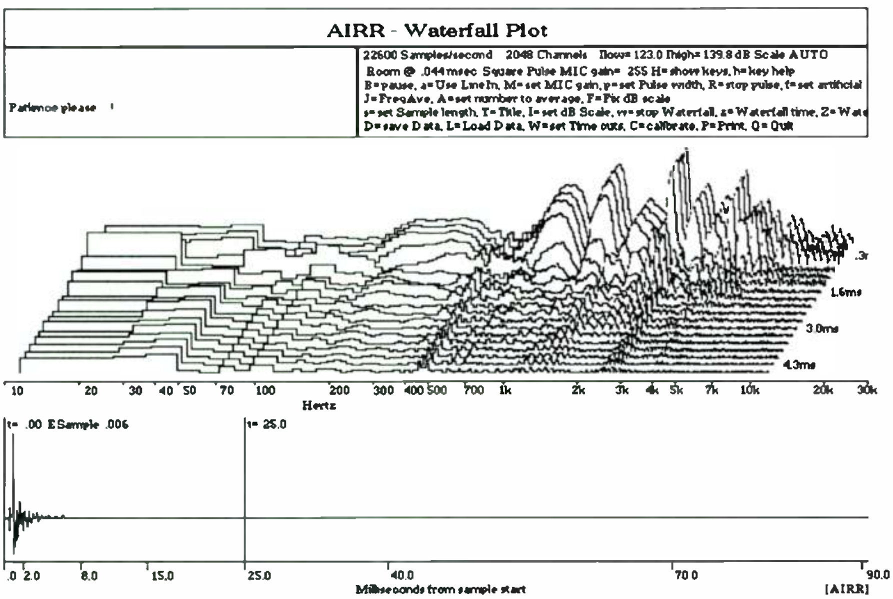


FIGURE 4: The waterfall plot.

DATA COLLECTION

To optimize the data collection rate, it is thus helpful to have a fast processor, fast memory (and plenty available), and the latest drivers from the sound card manufacturer. (Avoid loading the drivers "high" or using, for example, QEMM 7.0™ to load the driver. I have observed very poor collection rates under these conditions.)

In practice, on a 386DX PC with a 40MHz chip and 500K of spare memory, the rate normally achieved is in the region of 30,000 digitizations per second (a less recent version of the driver produced 24,000 per second), or, in other words, a maximum frequency point on the plots at 15kHz. On machines with faster CPUs (such as the 486 or Pentium series), this rate may be higher. (In the next version of AIRR, I intend to improve the data collection speed by using either assembler or DMA.)

You can check the accuracy of the AIRR calibration in one of several ways. The most obvious is to feed the sound card with a sine wave signal of precisely known frequency, such as that from a test CD or calibrated signal generator, and observe the position of the peak that results in the frequency plot. For this, AIRR is switched into no pulse and frequency averaging mode, and the signal sampled.

Note that AIRR must be in the frequency averaging mode, since averaging in the time domain would result in a uniformly flat signal. This is because there is no oscilloscope-like triggering employed on the input signal, so each sample would tend to begin at a different position on the waveform. To demonstrate this method, *Fig. 1* shows the result of frequency averaging the sound picked up from continuous playing of four sine wave tracks of different frequency from a test CD (the AIRR screen grabs have been converted from color to black and white for the purpose of including them in this article).

AIRR DISPLAY FEATURES

The screen is divided into five regions. The topmost box displays the title, which may be altered if desired. Beneath and to the left is an I/O box for brief information on the current status, and also as an input box when you're required to enter text or other data.

For example, when specifying the number of samples over which to average, a prompt appears in the box showing the minimum and maximum values allowed as input, together with the current value. An attempt to enter a number other than in the correct range results in a message to that effect, and a chance to try again. To the right of this is a status box detailing the current running conditions and showing the sample rate as calculated by the last calibration pass, the number of channels/digitizations that are currently being sampled as a block, and the calculated lower and upper limits of the amplitude in the time domain plot. The following line indicates the mode of operation: room response, amplifier response, no pulse or artificial feed.

In room mode, the gain being used for the MIC input on the sound card is shown. *[The AIRR program messages and prompts follow the standard abbreviation for microphone.* Speaker Builder *prefers the alternative style, mike.—Ed.]* For amplifier mode the gain on the line input is shown. Then, for the pulse modes of operation, the width of the pulse appears, followed by its type (square or sin(x)/x). Immediately below these two lines of information is a list of command letters that may be typed, with a brief description of the function of each.

Next down the screen is the region devoted to the frequency response plot. Both axes are plotted on logarithmic scales. The program continually adjusts the vertical axis towards showing the whole spectrum curve without clipping. For the waterfall plots, the vertical axis is not shown. Beneath the frequency plot appears the time domain plot. This shows the amplitude of the sample as a function of time in milliseconds from the start of the sample. The x-axis is relabelled whenever a calibration is performed.

Again, the vertical scale is continually adjusted so the whole signal appears in the plot. The positions of the time domain cut variables are shown as vertical lines and associated text; positioning the cuts is achieved by entering new values in the I/O box after selecting the appropriate option key. Also shown is the "sample energy," which is simply the integrated area beneath the sample. This is calculated by summing the absolute value of each digitization and dividing the result by the data rate; it gives some measure of the power in the pulse.

PULSE GENERATION & COLLECTION

A single pulse of minimal width is generated by sending a nonzero byte to the DAC on the sound board, immediately followed by a zero byte. This causes the analog signal level at the DAC output to go high, then low, reproducing a pulse of width typically in the region of 50ms, corresponding to a first pole in the frequency domain at 20kHz. The pulse

TABLE 1

FUNCTIONS (AND ASSOCIATED KEYS) SUPPORTED BY AIRR

TABLE 1
FUNCTIONS (AND ASSOCIATED KEYS) SUPPORTED BY AIRR

Description	Key	Input Required
Set the number of samples over which to average	A	Integer
Toggle between measuring signals from MIC or line-in socket	a	None
Freeze the display, and stop sampling until a key is pressed	B	None
Calibrate the AIRR data collection rate	C	None
Write current response data to disk	D	Name of file to store ASCII data
Toggle automatic or fixed scale in the frequency domain plot	F	None
With pulse generation, check what a perfect component would produce as a frequency response	f	None
Show this table	H	None
Select help on a key	h	None
Specify upper and lower limits on the vertical axis of the frequency domain plot	I	Upper and lower limits in dB
Toggle between averaging the time or frequency domains	J	None
Load response data from disk	L	Name of file with data
Set the level of the on-board amplifier for MIC or line inputs	M	Level from 0 to 255
Plot the screen into a .PCX file	P	Name of file to contain PCX data
Specify the width of pulses to send. Also used to change pulse type	p	Square pulse width (+ve values) or Sin(x)/x pulse width (-ve values)
Quit AIRR	Q	None
Toggle pulse generation on or off	R	None
Toggle the subtraction of the previously measured frequency response from the current response on or off	S	None
Set the sample length	s	Power of 2 in the range 4 to 12 (giving sample lengths from 16 to 4,096)
Change the title	T	New title
Show the AIRR version number	V	None
Specify the cut points in the time domain plot	W	Upper and lower cut point positions, in milliseconds
Toggle the waterfall plot off and on	w	None
Select the number of FFTs to cover the waterfall plot	Z	Number in range 2 to 100
Select the time duration of the waterfall plot	z	Time in milliseconds

begins its journey through the electronic components on the sound card, and then into the attached audio components under test.

Two possibilities exist: either the component under test is a loudspeaker and associated amplifier, or it is the amplifier itself. In the former case, the pulse travels through the amplifier circuitry, is reproduced as sound by the loudspeaker system, and then converted back to an electrical signal by the microphone, and fed back into the sound card via the line-in socket. *Figure 2* shows the standard setup for connecting the various components.

SIGNAL PROCESSING

The pulse that is picked up by the sound card is digitized into a series of 8-bit voltage levels. (The 8-bit mode is used in the current version for simplicity; my next version will benefit from 16-bit mode, although the increase in the resolution on the *magnitude* of the signal that this affords will not significantly affect the results.)

As already mentioned, the *number* of digitizations that are taken of the incoming signal and the *rate* at which they are gathered determine the lowest and highest frequency components that may be resolved. For example, if the sample rate is 20,000 digitizations per second, and 1,024 are taken, then we have a "window" on the arriving pulse that is about 1/20 of a second long, and so we can in principle resolve frequencies down to 40Hz.

In AIRR, you may select the number of digitizations that are taken to make up the window on the incoming pulse, with the restriction that this number is a power of two. This limitation simply arises from the number of data points that may be passed as a sample to the FFT. By default, AIRR uses blocks of 512, but offers a choice ranging between 16 to 4,096 in powers of two. The smaller the number of data points in the window, the quicker the data is collected, and the faster the turnaround occurs. Larger sample sizes afford lower frequency measurements.

Each sample block of N digitizations is passed through the FFT, which decomposes it into a set of amplitudes in N/2 bins of frequency. "Cuts" may be placed on the incoming signal, so that effectively only a portion of it is transformed. The amplitudes of the signal outside the cut limits are set to the quiescent value. By default, the cut lines are at T=0.0 and T=(Sample Length)/(Sample Rate), so the whole sample is within the cut.

Figure 3 shows the AIRR results for a signal arriving at the microphone input. Just before the pulse, an excursion from the quiescent level of the signal can be observed. This is due to crosstalk between the ADC and DAC signal paths on the sound card: it is the pickup of the outgoing pulse generated by the ADC, and is an ideal candidate for cutting with the cut lines. In this case you might specify a lower cut position at 1.0ms.

The upper cut line is for excluding the part of the sample starting from the first of the small pulses (such as wall reflections) that appear at the microphone. Making this upper cut allows AIRR to measure the anechoic response of the components under test. Without the upper cut, AIRR measures the so-called "in-room" response.

WATERFALLS

The waterfall plot is useful for observing the spectral response of the system under test as a function of time. It can show unwanted acoustic resonances in loudspeaker cabinets, for example. The waterfall plot is generated by taking the FFT of the whole time domain sample, then the whole sample after ΔT seconds have elapsed, then after twice ΔT seconds, and so on. Then the set of FFTs is plotted on a pseudo-3D plot with time along the z-axis. Waterfall plots have already been covered in *SB* (Bill Waslo, "Letters," reply to P.P.N Thompson, *SB* 1/94, p. 66).

AIRR offers a waterfall plot option that lets you specify the length of time the z-axis should span and the corresponding number of FFTs that should be taken over this time. The time is counted from the lower cut position, or zero, whichever is larger. The program defaults to plot 30 FFTs evenly spaced over 5ms. *Figure 4* shows the waterfall plot generated by AIRR for a loudspeaker system.

The disadvantage of the waterfall plot is the time it takes to calculate and display. Normally, you will only invoke the waterfall plot option once a satisfactory standard response plot has been obtained by averaging over many sound samples.

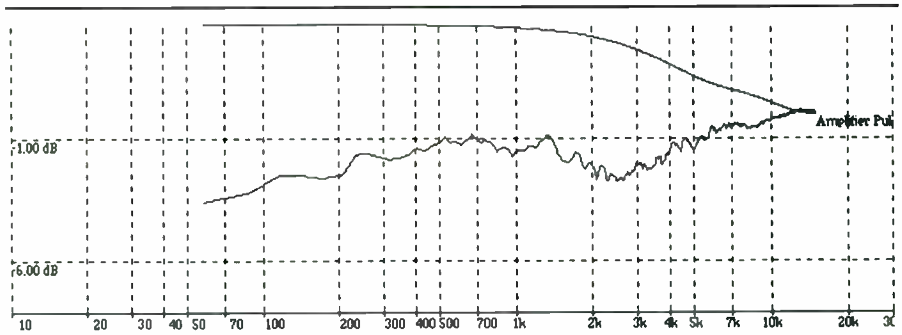


FIGURE 5: The smooth curve shows the response of the amplifier to the input pulse; the wobbly line is the response of the loudspeaker/amplifier combination, corrected by this curve.

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DUPONT

SUBTRACTING RESPONSE PLOTS

When measuring loudspeaker response, it is important to understand the extent the signal is being distorted before arriving at the loudspeaker. To do this, you need to measure the response of the system up to, but not including, the loudspeaker. Then you subtract that response curve from the response curve obtained for the whole system, loudspeaker included. This leaves the desired curve (which, however, is still affected by any nonlinear response due to the microphone).

The current response curve being measured can be adjusted in this way in AIRR. To correct for amplifier response, you first measure the amplifier response using the "line-in" source, then swap to measuring the response using the microphone, and then select the subtraction option, "S." *Figure 5* shows the result of subtracting an amplifier response curve from the loudspeaker curve in this way.

STORING SPECTRA ON DISK

Frequency response data may be stored on disk as ASCII text, which can then either be read back into AIRR at a later date or imported into another tool such as a spreadsheet. When a saved response curve is read back into AIRR, it appears in gray on the same axes as the current frequency response plot, so you can easily see the difference. If the data has been stored in the file at a different resolution (sample length) than currently in use, then the program performs either interpolation between existing data points or removal of excess data points.

The restored data may be subtracted from the current data if desired. This is useful, for example, if you have a set of data points that describe the response of the microphone being used. By keeping the data in a file you can, while measuring a loudspeaker response curve, load the microphone response and then subtract it from the curve.

MODIFYING PULSE SHAPE

The program offers a choice of two kinds of pulse—square or following sin(x)/x behavior. AIRR's default is a square-wave pulse of the shortest possible duration, in this case, the time to send the DAC high then low with two sequential high-level language instructions at maximum speed. In practice, this works out to be slightly faster than reading a sample (which also requires two sequential instructions, but to the ADC instead).

ACKNOWLEDGEMENTS

I am indebted to the staff from Creative Labs who answered some of my questions on CompuServe (GO CREATIVE) and provided the information I needed to program the SoundBlaster. In addition, I'd like to thank Ken Morse, a fellow CompuServe member with whom I bartered FFT code in exchange for an invaluable list of SoundBlaster command bytes.

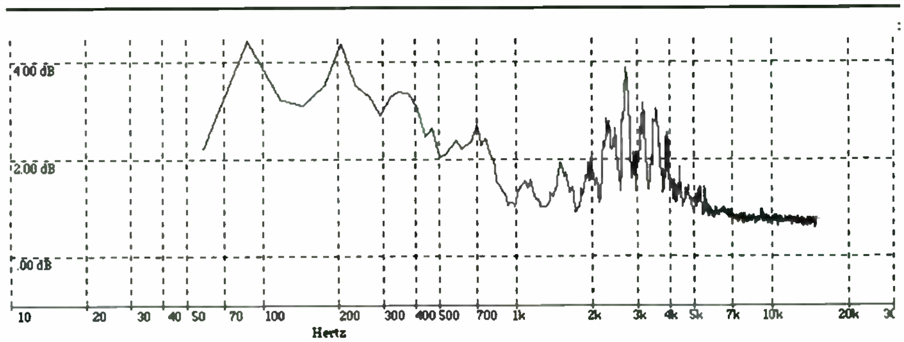


FIGURE 6: Frequency response for a loudspeaker system as measured using the microphone supplied with the SoundBlaster board.

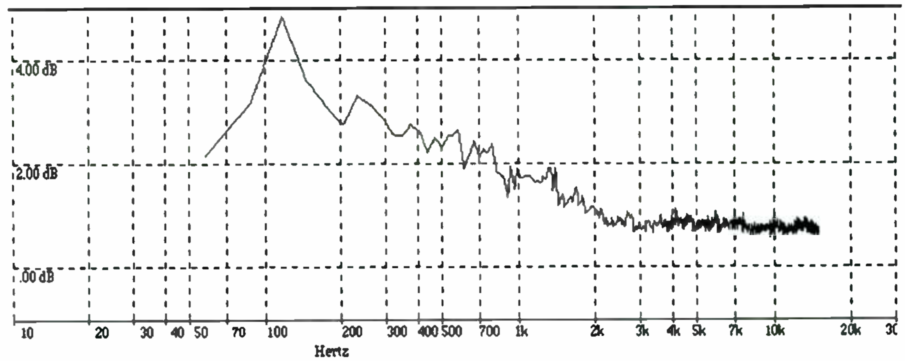


FIGURE 7: The response as measured using the Yamaha MM-110 microphone, for the same loudspeaker system.

Measurements indicate that the pulse duration obtained in this way is around two-thirds of the duration to read one ADC digitizing. To increase the width of the pulse sent by AIRR, press "p," then enter a number, N, which is used as a multiplier. The effect is to send the DAC to a preset level, the same level to the DAC N-1 times, then a zero level.

If you are unsure what the response of a distortion-free audio component would be to a wide pulse of this sort, then you can see by requesting AIRR to "send" the pulse directly to the FFT. This is achieved using "F" for artificial feed. As the width of the pulse sent increases, the first pole of the frequency response decreases in frequency. By experimenting with pulses of various widths, you will observe that the shape of the frequency response curve is that of sin(x)/x.

For pulse widths equal to the sample length itself, the sin(x)/x curve is very apparent. It turns out that the inverse FFT of a sin(x)/x signal is just a band-limited flat spectrum. This is the result of sin(x)/x pulses in telephony, where guaranteed bandwidth of digital signals is crucial.

AIRR's sin(x)/x pulse form causes a suitable set of levels to be sent to the DAC on the card. Due to the way the current version of AIRR works, and as with wide square pulses, the time for the first DAC-generated voltage excursion to travel through the system and be picked up by the ADC must be greater than the time to send the whole pulse shape to the DAC. Otherwise, the digitization of the signal will start too late to pick up the pulse start. This can be avoided by suitably reducing the width of the pulse.

MICROPHONE ISSUES

Microphone response clearly plays a crucial role in the accuracy of the spectra obtained using AIRR. A poor microphone (one with an uneven response over the range of frequencies to be measured) is useless for making measurements. You usually end up measuring the response curve of the microphone itself! The microphone packaged with my SoundBlaster board bears the inscription "CT329" and, just beneath, "Impedance 600Ω." No other documentation accompanied it.

Using a microphone of accurately known characteristics, such as the Mitey Mike, is clearly more advantageous. In this case, every spectrum obtained with AIRR using the Mitey Mike could be corrected on a point-by-point basis using a software lookup table. The best solution would, of course, be to measure the response of the microphone directly, using AIRR. I confess to having no idea how this might be achieved.

My Yamaha EQ-500 graphic equalizer came with a microphone for measuring and correcting in-room response (using a pink

This program is available as SOF-AIR1B36 for $49.95 plus $3 shipping in the USA from Old Colony Sound Lab, PO Box 243, Dept. B94, Peterborough, NH 03458.

noise generator). This microphone appears to be of good quality and is labelled "Electret Condenser Microphone MM-110." It is powered by a single AA-size battery. By using it instead of the SoundBlaster microphone to make some comparative measurements of a loudspeaker, I obtained the spectra shown in *Figs. 6* and *7*. (These spectra were for frequency averaged pink noise.) As you can see, there are significant differences!

ELECTRONICS QUALITY

Apart from the microphone, another aspect of the AIRR system which might introduce nonlinearities into the measurements is the quality of the electronics on the sound card itself. The frequency response of the on-board amplifiers, mixers, DAC, and ADC are all important.

Without professional-grade equipment to measure the response, I looked up a recent review of 24 sound cards published in *PC* magazine.[1] The review lists the recording and playback frequency response of each card. Both of the Creative Labs SoundBlaster cards showed a frequency response flat to within 1dB from 100Hz up to 20kHz. Signal-to-noise (S/N) ratios were in the region of 70dB, and total harmonic distortion (THD) around 0.75%. Incidentally, one of the SoundBlaster cards (Basic Edition) was awarded Editor's Choice.

CONCLUSION

The latest generation of sound cards for the PC offers you a sophisticated tool with which to develop audio signal processing software. The AIRR program profits from the programming interface to a SoundBlaster card by implementing a spectral analysis function with fast Fourier transforms.

The current version of AIRR supports many functions (*Table 1*), but the scope for improvement to the initial design of AIRR is considerable. My "to-do" list includes optimizing the execution speed, using DMA transfers rather than direct access commands, including phase response plots, and implementing a choice between the existing square and sin(x)/x pulses, MLS (à la IMP) and pink and white noise as the stimuli.

[Although Creative Labs' SoundBlaster card is popular, in its minimal forms it supports only a single brand of CD-ROM drive, its own. If you own a popular brand of CD-ROM, you will need to keep its interface card in your computer, in addition to the SoundBlaster card. If you buy SoundBlaster's more expensive "multi-CD" card, you can save a computer slot. MediaVision's 16-bit card is SoundBlaster-compatible—Ed.]

REFERENCES

1. John R. Quinn, "Big Audio Dynamite," *PC Magazine*, Vol. 13 No. 7, April 1994.

RESURRECTING A PAIR OF AR2AXs

By Hilary Paprocki

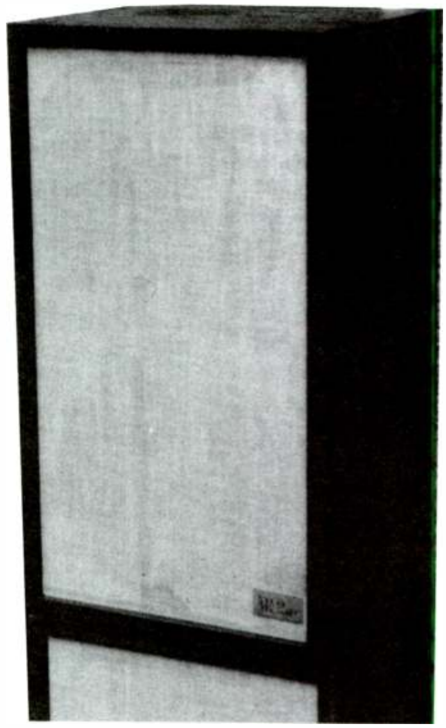


PHOTO 1: One of the treasures the author uncovered on the garage sale circuit last year was this AR2ax speaker.

I wouldn't want a Porsche. Honest. If you gave me one, I'd sell it today. The incremental value at those price levels is meaningless, compared to what you could do with the money. You could fly a plane. You could knock down a slum building and plant trees. You could bum the world for a year. If it's a car I need, show me an early Pinto with a 350 engine and we can talk.

With this kind of attitude, you can imagine that I get a kick out of finding old speakers at garage sales and determining what kind of performance can be squeezed from them. Last summer's big find was a usable table saw and a pair of Acoustic Research 2ax speakers, all for $75.

A HIP COMPANY

Younger readers should know about Acoustic Research. The company gets credit for inventing and commercializing sealed-box loudspeaker systems at the beginning of the '50s. At that time, a normal hi-fi speaker occupied about a cubic yard or more. And even at that size, the bass didn't really go very low. When the tiny AR-1 woofer system was introduced, at about 1.5 ft^3, it changed the world.

AR was one of the first successful companies in hi-fi history to fine-tune their products primarily by ear. This caused their products to stand out from the norm in all kinds of ways. When everyone else was using electrolytic capacitors in their crossovers, AR used special film caps. The midrange driver's low-pass filter was a slice of fiberglass wool in front of the cone. The company provided extra wire terminals for biamplifying.

When other companies advertised frequency-response curves, AR was holding live-versus-recorded demonstrations. We're talking 40 years ago here. AR was a very, very hip company, and led to what we now call the high-end in methods and philosophies.

HI-FI HOT ROD

As a result of my restoration project, the AR2ax speakers are making very acceptable sounds in my bedroom right now. If you enjoy building hi-fi hot rods, these classic speakers make a fine project. In case you encounter a pair of boxes you'd like to play with, I'd like to share with you some of my observations during the restoration, in a list, in no particular order.

1. The woofers had foam suspensions, and as you can imagine, the 25-year-old foam was crumbling. The drivers can be refoamed, which means that the suspensions can be replaced. Pro-Cone in Rochester (716-328-9770) did an amazing job with mine. You can't tell that the speakers aren't brand new.

2. My pair included one blown tweeter, but what seems like a blown tweeter in yours might just be a bad level control. The AR2ax has controls on both the tweeter and midrange drivers. The control contacts often corrode. Use the remaining good controls to find the right tonal balance, measure the resistances among the three pot terminals, and replace the controls with good 5W fixed resistors.

While you're soldering, notice that when you replace the tweeter and midrange level pots with resistors, you end up with one resistor across each driver, and then one resistor in series with that combination, going to the input terminals. There's no reason not to locate these resistors up by the drivers. That way, when you perform final tuning, you don't need to repeatedly haul all that fiberglass in and out of the cabinets. Just pop out the midrange or tweeter, and its level-control resistors will be right there (*Fig. 1*).

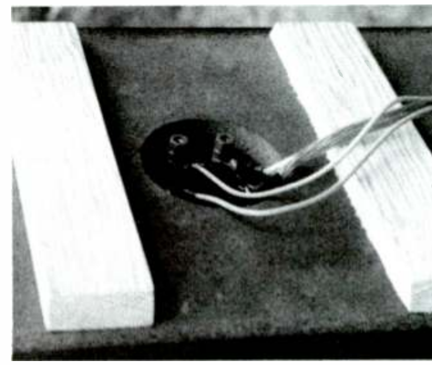


PHOTO 2: The speaker back panel showing the biwiring and the "minimum-effort, minimum-intrusion" braces.

Of course, if you don't mind the exposed clutter, you can just build the crossovers outside the box. There are sonic advantages to doing it this way. Some crossover components are microphonic and really shouldn't be in the box.

3. If your blown tweeter really is, then you'll need to perform a quick repair. The old capacitors may short or change value. If this allows low frequencies into the tweeters, they'll pop. To be safe, replace the original capacitors with good Mylar® (or fancier) new ones. I used Mylar with 0.1μF polypropylene bypasses in parallel, because that's what I had lying around.

4. And while you're at that, you can give the speaker a faster, more agile, more detailed sound by lowering the woofer/midrange crossover frequency just a tiny bit. What was good sound according to '50s' standards is considered sluggish and slow today.

Add 1μF to the original 6μF midrange capacitor value, and five turns of wire to the woofer's crossover inductor coil. It's easy to do, even with the coil inside the box. Just use some 18GA enameled wire. Feed the AR inductor wire (the end that went to the woofer) back through the hole in the flange of the bobbin so it's free, and solder on a length of new wire. As you add the five turns, pull the wire so that it is good and tight and bring the new wire through the flange hole and up so you can solder it right to the woofer terminal (*Fig. 2*). This will decrease your midrange

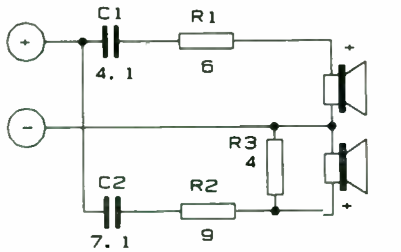


FIGURE 1: Schematic of the author's speaker system, after replacing the pots.

driver's power handling capacity, but unless you routinely blow speakers, don't worry about it. Mine are still fine.

Incidentally, it's amazing how sensitive this adjustment is. Only three turns add a woofy boom to many recorded voices (especially Louis Prima, and you certainly don't want to shortchange him). At six turns you begin to move toward an artificially detailed hi-fi-microscope sound. If you're dealing with these kinds of sonic qualities in your homemade system, try tinkering with turn counts on the woofer inductor.

For reference, I made these observations with a quad-regulated Dynaco Stereo 70 amplifier with a mostly stock circuit and GE 6CA7s. I know they're not hip tubes, but I was going for durability.

5. I bought a new AR tweeter to replace the blown one in my pair. It cost a million dollars, and the remaining old one failed soon after. Maybe the tweeters deteriorate over time too. Maybe you should consider buying new modern tweeters from one of the *Speaker Builder* advertisers.

I sent letters to several of them asking what would be a good replacement for the tweeters in this particular three-way speaker. One company sent a scrawl that I couldn't read. Another company sent a stack of photocopied pages that must have cost a fortune, but didn't recommend any of the products described. Another company recommended a midrange driver. I recommend you don't bother asking. Just find out what one of the similar good commercial systems uses, and get those.

6. You *will* use solder joints instead of slip-on terminals for all the drivers and components, won't you? Use nice wire inside the boxes, too, such as the solid 18GA pure cop-

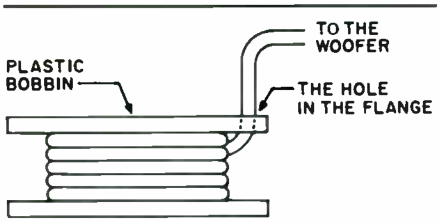
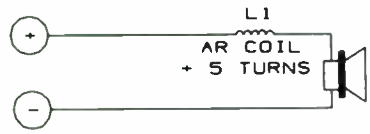


FIGURE 2: The AR woofer inductor.



per hookup wire at Radio Shack or the enameled stuff you bought for the crossover.

7. Notice also that the high-frequency and midrange speakers are out of phase with the woofer as the speaker was originally built. That did produce a nice simulated depth in the sound, but if you switch them around so they're back in phase, you can achieve surprisingly accurate instrumental tones.

8. Of course, you may wish to experiment with this. Since you won't be using the level controls, you now have more than enough holes in the terminal board to mount two pairs of binding posts—one pair for the woofer and one isolated pair for the mid-tweet bank. This lets you switch around the mid-high phase relative to the woofer, plus you can now use separate amplifier cables for the woofer and the mid-tweet, individually chosen for their respective relevant qualities.

9. I'm using some old Monster Cable for the woofer and two runs of Belden 1242A telephone wire (one four-wire cable hot, one four-wire cable common) for the uppers, crimped together into one pair of spade lugs where they meet the amp.

Imagine a fancy speaker cable consisting of eight strands of 99.9999% pure copper, each strand individually insulated in a special low-dielectric absorption polyethylene material. Sound like nine bucks a foot? Spend nine cents and get 1242A.

10. Don't forget to put a little epoxy on the terminal posts when you mount them on the terminal plate to keep them from spinning when you tighten the speaker cable ends onto them.

11. And while you're in there, why not brace the cabinet? I used two 3/8" × 1" ribs across the baffle inside, glued on edge between the woofer and the mid-tweet holes. Cutting another baffle board to double-thick the original is even better, if you're not going to use the grilles. The heavier the baffle the better. On the back, I added two 3" × ¾" fir ribs flat to the outside. With beveled ends, they aren't visible from the front. Don't stuff the box full of wood; you need air in there (*Fig. 3*).

12. The blue temporary gasket silicon sold by the auto parts store (Permatex or Prime Seal) makes a great sealer for the drivers when you mount them to the baffle.

13. This is wacky, but it works: buy some rubber cement and a bag of cotton balls at your local drugstore. Arrange the cotton balls in a circle on the tweeter and midrange mounting plates and rubber-cement the balls in place. This prevents uncontrolled diffraction and does wonders for the focus and smoothness of your sound. They'll come off easily if you don't like them (*Fig. 4*).

14. The cabinet finish should not be a big deal, unless you found a pair in fabulous '50s limed oak. If that's the case, preserve them as well as you can. I suggest you avoid silicone polishes, which in the long term can have regrettable effects.

If your boxes are the usual oiled walnut, sand 'em down with 120, then 220, in a sanding block. If some scratches remain, ignore them. Oil the veneer with linseed oil, thinned with turpentine, on a pad of fine steel wool. The old-timers suggest that you oil wood every day for a week, every week for a month, and every month for a year. Just make sure the oil is well-thinned, and keep the steel wool hairs way away from the drivers.

15. Concrete blocks are okay for stands, but fix them so they don't rock. A cheap masonry bit will zoom right through a cinderblock, allowing you to install plastic anchors and lag screws to act as feet—three top and three bottom (tripods, as you know, don't rock). Watch out for scratching your wood, though. You could use the holes to install unfinished wood drawer knobs, which work fine and won't mar hard floors.

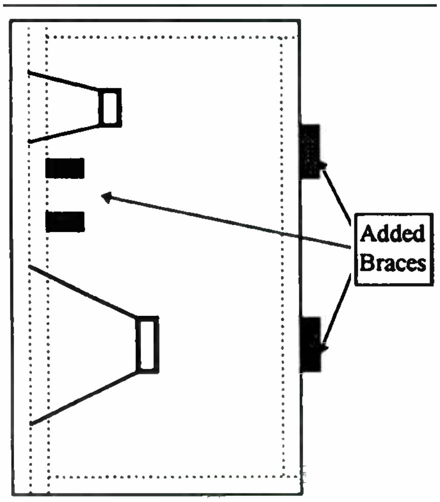


FIGURE 3: Bracing the inside and back of the cabinet.

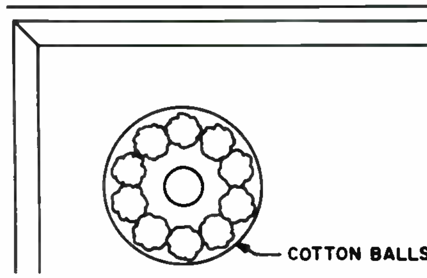


FIGURE 4: According to the author, adding cotton balls to the tweeter and midrange can improve the sound of your system.

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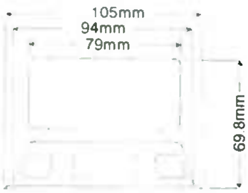
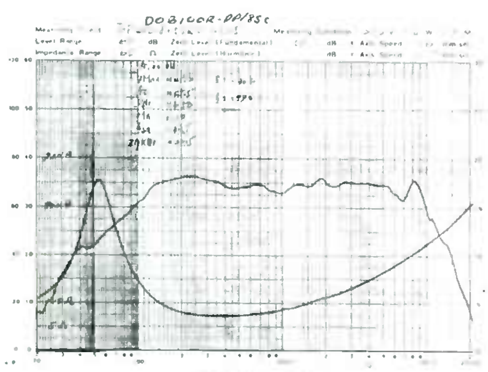
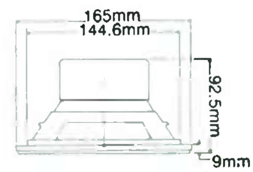


GENERAL INFORMATION			
Impedance	Z	8.0	Ohms
D.C. Resistance	Rdc	6.3	Ohms
Resonance Freq.	Fs	65.0	Hz
Characteristic Efficiency Level	SPL	85.0	dB1W/1M
Power Handling		60.0	*W
Upper Frequency		10,000	Hz
Cone Material	Carbon Fiber Polypropylene		
Surround Material			Rubber
Total Weight		0.940	Kg
PARAMETERS			
Mechanical Q Factor	Qms	1.88	
Electrical Q Factor	Qes	0.43	
Total Q Factor	Qts	0.35	
Equivalent Volume	Vas	6.3	L
VOICE COIL			
Xmax		2.0	mm
Diameter	d	25.5	mm
Winding Length	h	10.0	mm
Layers	n	2.0	
Former Material			Aluminum
Height of Gap	hg	6.0	mm
Inductance	L	0.56	mh
MAGNET			
Weight		.226+.150	Kg
Flux in Gap	0	0.505	mWb
Flux Density	B	1.05	T
Force Factor	BL	4.98	N/A
SUSPENSION			
Moving Mass	Mms	5.73	g
Mechanical Resistance	Rms	1.065	Kg/s
Suspension Compliance	Cms	1.435	mm/N
Emissive Diameter of Diaphragm	D	8.416	cm
Effective Piston Area	Sd	55.63	cm2

*Music Power

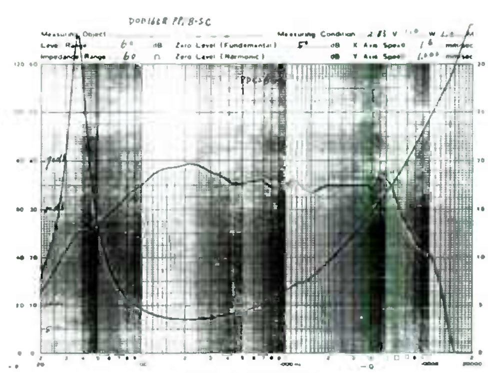


6 1/2" DOB165R-PP/8SC



GENERAL INFORMATION			
Impedance	Z	8.0	Ohms
D.C. Resistance	Rdc	6.3	Ohms
Resonance Freq.	Fs	36.0	Hz
Characteristic Efficiency Level	SPL	86.0	dB1W/1M
Power Handling		80.0	*W
Upper Frequency		6,000	Hz
Cone Material	Carbon Fiber Polypropylene		
Surround Material			Rubber
Total Weight		1.020	Kg
PARAMETERS			
Mechanical Q Factor	Qms	3.49	
Electrical Q Factor	Qes	0.33	
Total Q Factor	Qts	0.30	
Equivalent Volume	Vas	35.28	L
VOICE COIL			
Xmax		1.75	mm
Diameter	d	25.5	mm
Winding Length	h	9.5	mm
Layers	n	4.0	
Former Material			Aluminum
Height of Gap	hg	6.0	mm
Inductance	L	1.437	mh
MAGNET			
Weight		.226+.150	Kg
Flux in Gap	0	0.394	mWb
Flux Density	B	0.82	T
Force Factor	BL	7.44	N/A
SUSPENSION			
Moving Mass	Mms	12.65	g
Mechanical Resistance	Rms	0.834	Kg/s
Suspension Compliance	Cms	1.493	mm/N
Emissive Diameter of Diaphragm	D	12.82	cm
Effective Piston Area	Sd	129.06	cm2

*Music Power



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ROUND THE HORN

By Philip Newell and Keith Holland

Together with higher efficiency and far-field propagation qualities, a horn's generally higher output capability allows it to still play a role in studio monitoring. The debate about horns versus direct radiators for such purposes has continued for decades. Few horns have been developed solely for studio use, thus most studio systems using horns have "borrowed" technology from the worlds of cinema, sound reinforcement, and public address.

At the Institute of Sound and Vibration Research [ISVR] at Southampton University, England, Keith Holland engaged in a five-year study to discover whether there was further potential for the studio use of midrange horns. One of the main objectives was to determine whether some of their less favorable characteristics were a function of horns *per se* or were inherited as aspects of the borrowed nature of the technology.

This article discusses the findings of the above research program, details aspects of horn performance considered undesirable for studio purposes, and separates the individual physical parameters of horn design which give rise to many unwanted acoustic properties. The conclusion attempts to define the limits of horn performance within which the greatest number of unpleasant sonic attributes can be eliminated. The work was based on physical and mathematical analysis of the problems, closely related to a rigorous series of listening tests.

WHAT HORNS DO

Horn use for public address and cinema loudspeaker applications is almost universally accepted as good practice. Indeed in many of these instances, no viable alternative exists, since some requirements, such as high electroacoustic efficiency and flexibility of directivity control, are only easily achieved by horns. Changes in music recording techniques have resulted in a tendency towards larger control rooms.

Concurrently, control room acoustic-design philosophy tends towards lower reverberation times, and, in many, true reverberation does not exist at all. In these relatively large and acoustically "dead" control rooms, a borderline case has been reached between studio monitor systems and small public address/sound reinforcement systems. Numerous designers have attempted to develop direct-radiator technology to meet these needs, but the many fine existing systems usually require very high amplifier power, and live much of their working lives close to their power-handling performance limits.

On the other hand, systems using horn-loaded midrange systems, even in very large control rooms of $60m^2$ or more, are rarely driven much beyond 20% of their design power-handling capacity. So they exhibit a long and stable working life, together with lower amplifier power requirements and a good reserve of damage tolerance. Another desirable attribute of horn loudspeakers is their tendency to produce an output in the form of a spherical expanding wave, free of many of the lobing problems of the pistonically derived output from direct radiators.

No one type of drive system claims overall superiority here, since many subjective aspects of performance differ greatly from one listener to another. However, if some of the negative attributes of one system can be detected, isolated, and ultimately circumvented, then it will provide designers of future systems with more options in their quest for optimum design requirements.

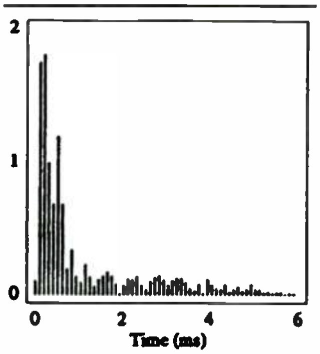

FIGURE 1A: Quad Electrostatic—original type.

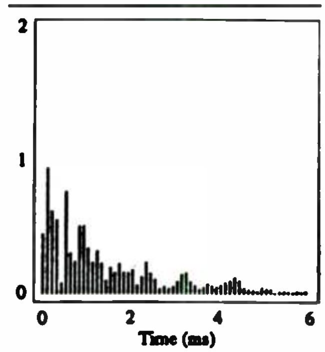

FIGURE 1B: Son Audax PR17-HR100-1AK7 cone midrange driver.

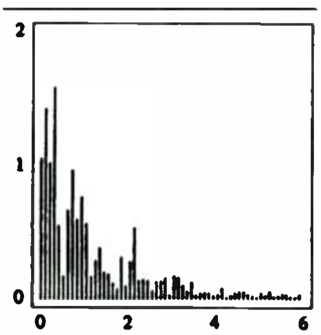

FIGURE 1C: Fostex H351-HA21 horn with Emilar compression driver.

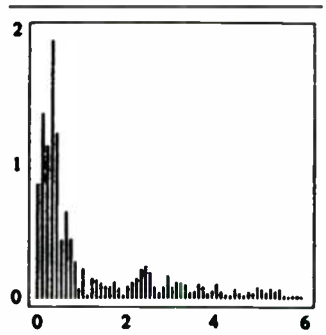

FIGURE 1D: High-frequency section of Tannoy 15" dual concentric.

This article first appeared in Studio Sound, *March 1994, pp. 59-70, and is reprinted here with kind permission from Spotlight Publications, London.*

TEST PROGRAM

The basis of the research work was to find links between measurable characteristics of horn performance and perceived subjective sonic characteristics. To reduce some of the tedium and general impracticability of first manufacturing, then setting up experiments to measure each interesting development suggested by the research, a one-parameter computer analysis technique was developed, then rigorously tested against actual measurements of real horns. This technique has already been published,[1,2] together with the development of an acoustic impedance tube measurement system,[3] which made practical the rapid measurements for the physical/numerical cross correlations.

An extensive program of listening tests was carried out over a six-month period in the large anechoic chamber of the ISVR, and again, a general outline of this procedure has been published.[4] Measurements of nonlinearities, both in the horn flares themselves and in the horn/driver combinations, were cross-referenced with the results of the listening tests, and a finite amplitude model was developed to predict nonlinearities in different flare shapes.[5,6] Cepstral analysis was undertaken to isolate the different horns' discrete reflection patterns not easily discernible from conventional measurement techniques.[7]

Holland compiled the research project into a thesis, for which he received his PhD at Southampton University in December 1992.[8] Since the individual aspects of the research are so well-documented in the references above, this article concentrates on the less widely promulgated aspects of the conclusions.

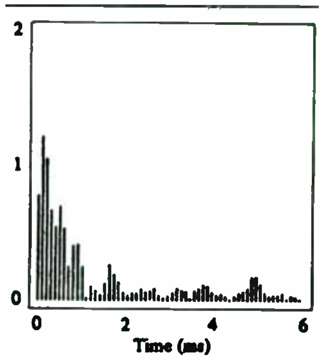

FIGURE 2A: Metal horn with Emilar EK175 compression driver.

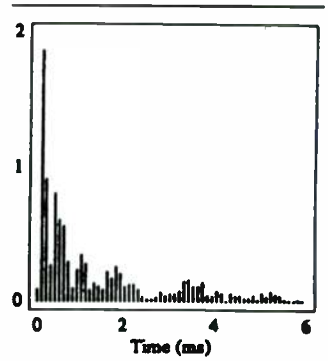

FIGURE 2B: JBL 2105 cone driver.

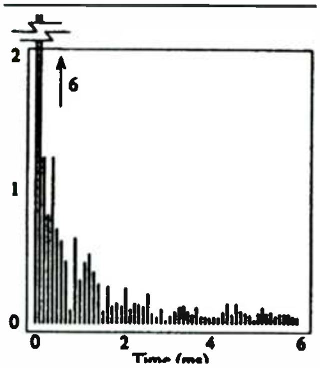

FIGURE 2C: JBL 2121 cone driver.

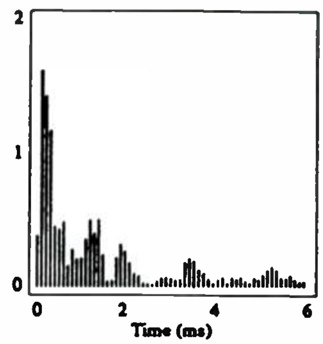

FIGURE 2D: AX1 axisymmetric horn with EK175 driver.

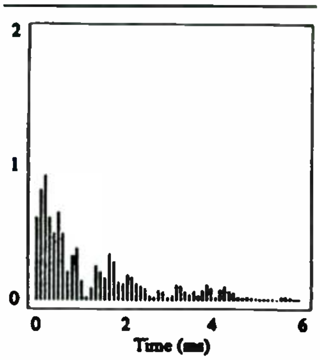

FIGURE 2E: Reflexion Arts horn with EK175 driver.

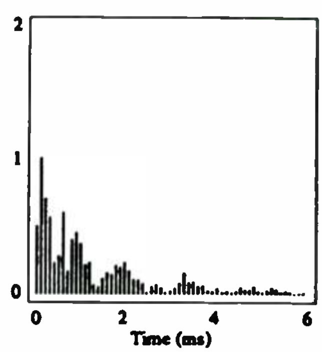

FIGURE 2F: Son Audax PR17-HR100-1AK7 cone midrange driver.

FOUR GROUPS

The listening tests involved over 7,000 comparisons of 20 different drive units and nine different sounds. The drive units consisted of horns of many different flare shapes, moving coil direct radiators, and an electrostatic. The nine sounds were essentially nonmusical, but contained different combinations of transient and steady-state or tonal content. They were band-limited 1–6kHz on playback, to avoid problems solely due to horn cutoff or other out-of-band anomalies.

It was hoped that the different combinations would help to isolate any "horn sound" which the units may possess. The initial question was whether all the horns would group together, either on some of the sounds or on all of them. After the results had been numerically, statistically, and "common sense" analyzed, there *were* groupings, but not in ways which had been anticipated.

Of the four reference "archetypes" to which other samples were compared, two were direct radiators and two were horns. The direct radiators included a Quad Electrostatic (*Fig. 1A*) and a Son Audax 6½" moving coil unit (*B*). The two horn archetypes were a Fostex H351/HA21 long (490mm) sectoral horn driven by an Emilar EK175 drive unit (*C*) and the "high" frequency section of a Tannoy 15" dual concentric, with the bass cone forming the high-frequency horn flare (*D*).

FINDINGS

In general, the horns with a throat-to-mouth distance of more than about 350mm were deemed to sound like the Fostex horn, while the horns with a throat-to-mouth distance of less than 350mm were generally judged to sound like the Son Audax direct radiator cone. Within each of the long/short groups, however, there were some odd exceptions.

One of the "long" horns was a Fostex wooden-flared sectoral horn, measuring 440mm from throat to mouth. Possibly more than any other horn, this one sounded like the Son Audax direct radiator cone. It consisted of a very short "throat extension," coupled to what were effectively large semicircular lips with a horizontal flare of 140°. The horn produced undesirable throat impedance plots, suggesting an uneven pressure amplitude response when connected to a driver. Yet, in auditioning before the tests began, the horn was generally considered "musical," pleasant to listen to, and most definitely not hornlike.

Among the short horns, 350mm or less, two failed to group with the direct radiator. One was the AX2, specifically designed for the tests as a result of the "seminumerical" one-parameter modeling and information gleaned from the use of the impedance tube. The horn was 180mm from throat to mouth and possessed a rapid flare which prior analysis had suggested as a requirement for a desirably smooth mouth termination, avoiding any sudden cross-sectional changes where the mouth meets the baffle. The horn showed an overall similarity to the Tannoy dual concentric, which was also a short, axisymmetric horn.

The third nongrouping horn was a Yamaha aluminum sectoral horn, which was a "borderline" 300mm in length. Not surprisingly, this horn straddled groups *B* and *C* in *Fig. 1*, but also gave many results dissimilar to any of the archetypes. Only on four of the nine sounds did it clearly group with *B* or *C*.

NONLINEAR DISTORTION

The first attempts to explain these anomalies—and the "break" at 300mm to 350mm—focused on nonlinear distortions. Much has been presented on the subject of horn distortion, to the extent that some circles believe that harmonic distortion makes many horns sound hard and unpleasant. Three predominating sources of this harmonic distortion exist:

1. electromechanical limitations of the drive unit, including thermal power compression effects, suspension nonlinearities, and magnet/gap problems;

2. nonlinearity produced as a function of the volumetric changes between the diaphragm and the phase plug on positive and negative half-cycles;

3. distortion produced by nonlinear propagation within the horn itself, which can, at very high levels, lead to shock formation.[9]

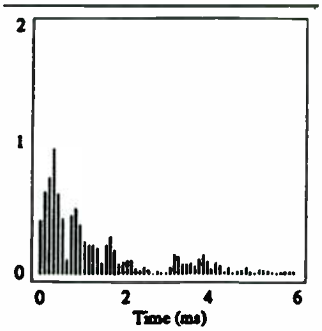

FIGURE 2G: Reflexion Arts horn with EK175 driver with lips sawed off.

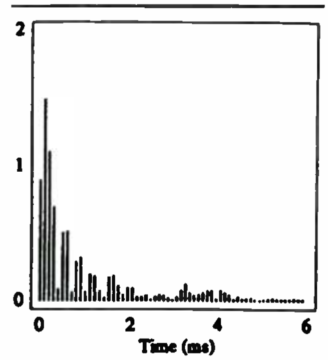

FIGURE 2H: AX2 axisymmetric horn.

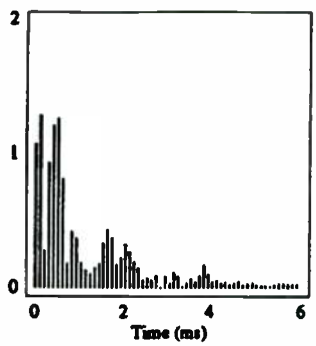

FIGURE 2I: Yamaha horn with EK175 driver.

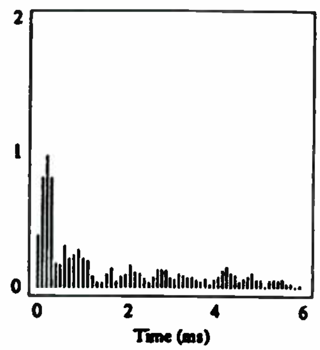

FIGURE 2J: Fostex H320 wooden radial horn with EK175 driver.

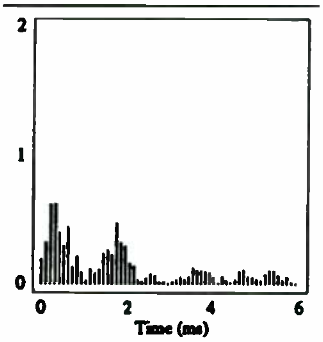

FIGURE 2K: JBL 2307-2308 horn/slant plate combination with EK175 driver.

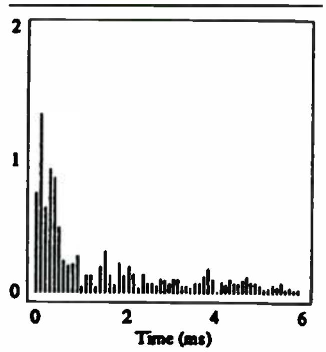

FIGURE 2L: Altec sectoral horn with EK175 driver.

TABLE 1

MEASURED HARMONIC DISTORTION LEVELS OF DRIVERS UNDER TEST

P(3m) = 75dB SPL

L.S.	f (Hz)	Input (v)	2nd	3rd	4th	5th
QUAD	1K	2.04	-59	-60	-69	-66
	2.8K	5.6	-62	-49	-	-
	5K	3.5	-66	-51	-	-
AUDAX	1K	1.01	-59	-54	-68	-
	2.8K	0.75	-61	-60	-	-
	5K	0.64	-62	-59	-	-
JBL	1K	0.2	-53	-75	-77	-
	2.8K	0.18	-38	-27	-46	-38
	5K	0.19	-47	-31	-59	-
EMILAR	1K	0.24	-61	-66	-	-
	2.8K	0.17	-58	-73	-70	-
	5K	0.3	-47	-57	-	-

P(3m) = 85dB SPL

L.S.	f (Hz)	Input (v)	2nd	3rd	4th	5th
QUAD	1K	6.02	-57	-57	-80	-
AUDAX	1K	3.8	-54	-49	-78	-67
	2.8K	1.7	-57	-57	-	-
	5K	2.5	-47	-67	-	-
JBL	1K	0.67	-42	-44	-60	-59
	2.8K	0.44	-43	-36	-49	-
	5K	0.54	-41	-37	-61	-
EMILAR	1K	0.73	-50	-55	-76	-
	2.8K	0.44	-49	-65	-73	-
	5K	0.87	-38	-54	-71	-

P(3m) = 95dB SPL

L.S.	f(Hz)	Input (v)	2nd	3rd	4th	5th
AUDAX	1K	10.1	-50	-44	-79	-69
	2.8K	5.7	-49	-49	-	-
	5K	7.7	-37	-57	-	-
JBL	1K	2.0	-41	-49	-55	-53
	2.8K	1.5	-35	-53	-67	-53
	5K	2.1	-29	-55	-	-
EMILAR	1K	2.45	-40	-53	-68	-59
	2.8K	1.4	-40	-55	-61	-63
	5K	2.9	-29	-48	-71	-

P(3m) = 105dB SPL

L.S.	f(Hz)	Input (v)	2nd	3rd	4th	5th
JBL	1K	7.8	-26	-41	-58	-51
	2.8K	4.4	-27	-43	-55	-57
	5K	7.1	-19	-37	-60	-
EMILAR	1K	10.5	-28	-32	-45	-55
	2.8K	4.4	-29	-52	-57	-59
	5K	8.8	-20	-36	-51	-

To test the less well-documented third cause, a finite amplitude model was devised for computer prediction. Most standard horn equations are calculated on the basis of infinitesimal wave amplitude, but in reality, usable sound waves have finite amplitudes. Superimposed on the initial sound waves are reflections from the mouth and obstructions within certain horns, plus complications due to phase dispersion within the horn flare.

The model proved highly successful and gave good correlation with measured results, which used a Community M4 as a signal source (capable of producing signals with less than 1% harmonic distortion, even at 150dB). The complicated and unwieldy test setup required two specially treated adjoining rooms, so once adequate verification of the computer model had been achieved, the computer model was certainly the most practical choice for further study on other horns.

MEASUREMENTS

The measured results (*Table 1*)—comparing harmonic distortion levels of a direct radiating cone, an electrostatic, and two horn loudspeakers, all used in the listening tests—show that at low levels (below 90dB) at 3m, there is no significant difference in distortion levels of the different devices. At very high levels, e.g., >110dB, very few drivers can produce such continuous sine wave levels, so comparison is not really relevant. Furthermore, certain audiological reasons also reduce the relevance of very high-level measurements in a studio environment.

From the above measurements and the computer analysis of the finite amplitude model, it was possible to attribute the distortions to one of the three main causes previously mentioned. Much second harmonic distortion can be attributed to propagation nonlinearities, with most higher-order harmonics being driver-related. None of this, however, falls into any sort of pattern when cross-correlated with the similarities and groupings in the listening tests.

Indeed, whichever ways the results were dissected and analyzed, no link could be demonstrated between harmonic distortion and audible similarity. Some units with up to 20dB difference in distortion levels sounded similar, while others of almost equal distortion figures sounded totally different. From the

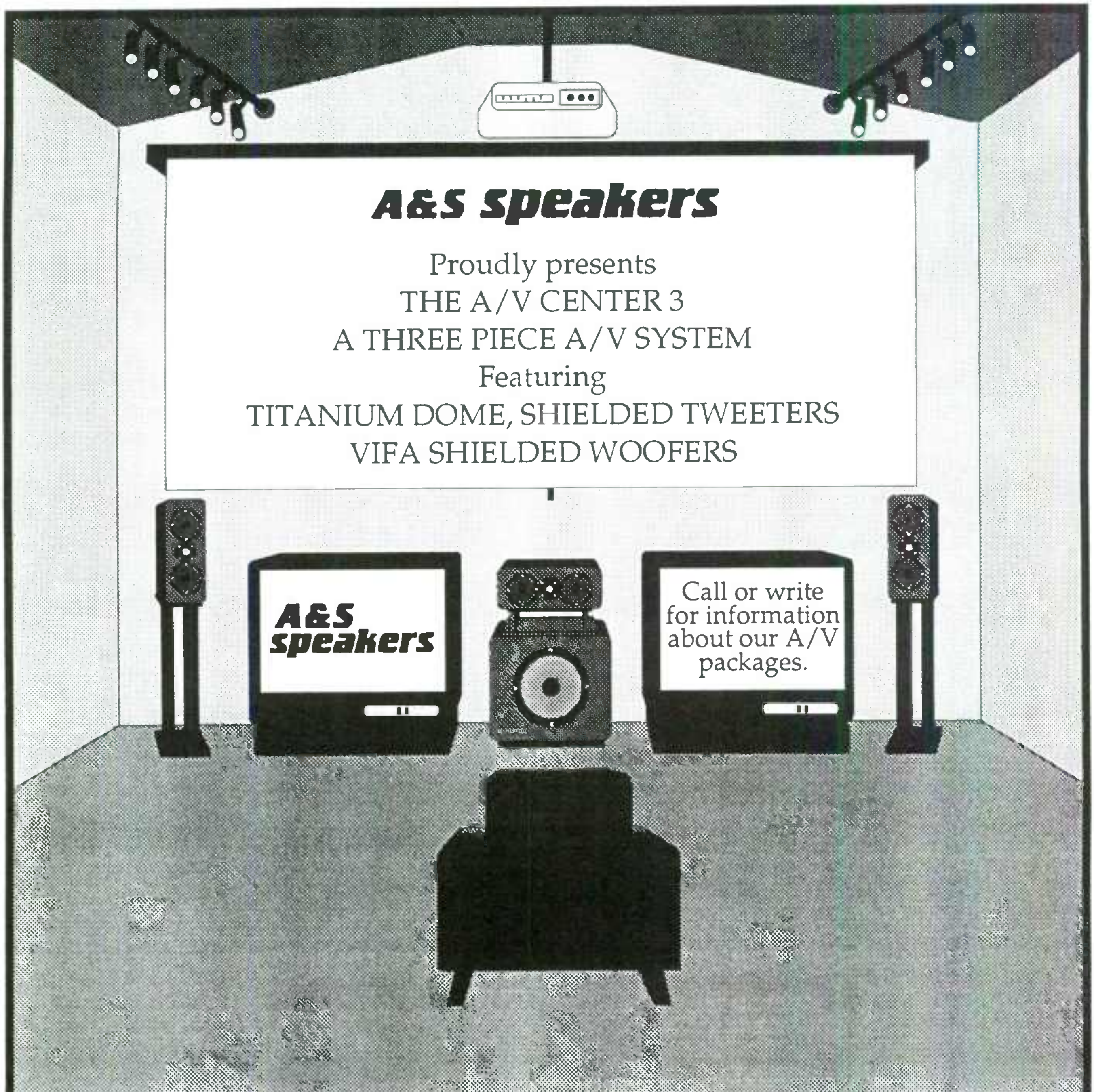

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PHOTO 1: Discossette Studio 3 in Lisbon, Portugal.

PHOTO 2: The Namouche Studio in Lisbon.

results of these tests and analyses, nonlinear distortions were most emphatically *not* responsible for any characteristic horn sound.

AMPLITUDE AND PHASE

Pressure amplitude response (frequency response) was another prime candidate for producing sonic similarity or dissimilarity. After the tests were completed, a Waveform Spectral Similarity index was calculated for each loudspeaker on each sound. This was derived by calculating the root-mean-squared error between the spectra of the original signal and that radiated by each loudspeaker.

Comparisons were made for speaker-to-filter input and speaker-to-speaker for each sound. A reasonably good connection was achieved here (around 80% similarity) between the calculated waveform similarity and the listening tests. Unfortunately, some results refused to correlate, in a most glaring way. Usually, when a sample driver which was deemed to be sonically similar to an archetype failed to show a similar pressure amplitude response, then a strong similarity was noted in the phase response. This has so far not been adequately explained.

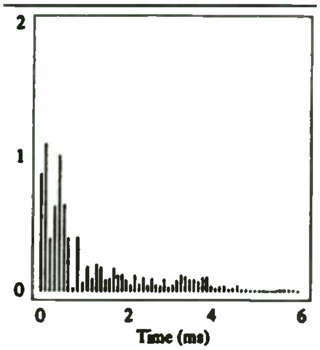

FIGURE 2M: Altec 806C multicellular horn with EK175 driver.

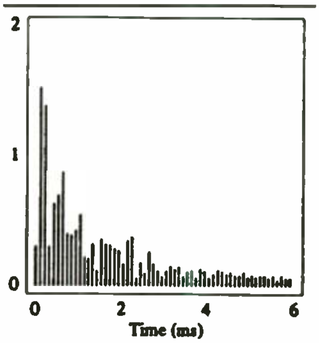

FIGURE 2N: Starr Singing Throat wooden gramophone horn with EK175 driver.

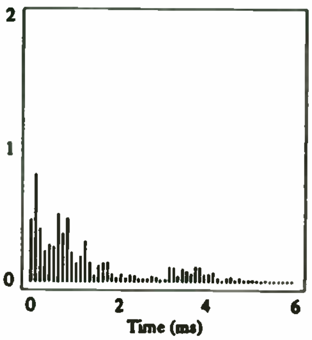

FIGURE 2O: Vitavox sectoral horn with EK175 driver.

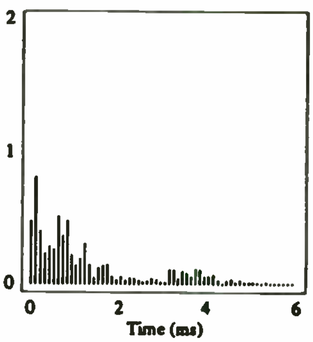

FIGURE 2P: JBL 2370 with JBL 2426 driver.

Certainly, the agreement between the listening test results and the comparisons between the spectra of the reproduced signals indicates that much of the acoustic similarity was due to the on-axis amplitude frequency response, but clearly, this was not the sole reason.

For example, a JBL 2370/2426 combination was very similar in its waveform spectral similarity to the Son Audax cone driver for all nine of the test signals. Yet in the listening tests it showed a reasonable similarity on only one of the nine sounds. It closely resembled the Fostex horn on five of the signals, and was judged similar to none of the archetypes on the other three signals. The phase response of the JBL combination was more similar to the two horns (archetypes *C* and *D*).

CEPSTRAL ANALYSIS

With neither amplitude, phase, nor harmonic distortions clearly explaining the sonic similarities or otherwise of the different drivers, it was decided to make further studies in the time domain. To further identify any reflections that may be produced at the mouth or within the flare of a horn, a form of power cepstrum was calculated from the modulus of the measured throat impedance. In this type of analysis, the frequency-domain representation of the throat impedance's modulus is treated as a spectrum; the power cepstrum is then calculated using Fourier transforms.

Cepstral analysis was first defined in the mid-1960s to help separate echoes from "clutter" in seismic research. The power cepstrum of a transfer function is the Fourier transform of the log of the transfer function's amplitude. Each driver's power cepstrum was plotted using a y-axis scaled in nondimensional decibels and an x-axis plotted in terms of both time and distance.

Figure 2 shows the power cepstra of the units used in the tests. The plots proved to be very revealing, since they are very effective in showing reflections. In a conventional pressure amplitude plot, a reflection would appear as a comb-filtering effect, but on a complex spectrum, this can be difficult to recognize. On a power cepstrum, however, reflections exist as single spikes along the time/distance axis, and can thus readily be recognized.

In general, the cepstral analysis demonstrated that the audible similarity groupings from the listening tests could be described in terms of the reflection patterns shown in the power cepstra. The various reflections and resonances produced in the cone of a direct radiating loudspeaker can give rise to irregularities in the frequency response function that are similar to those due to mouth reflections in short (sub 350mm) horns. This explained the anomalous behavior of the "long" Fostex wooden horn in the listening tests.

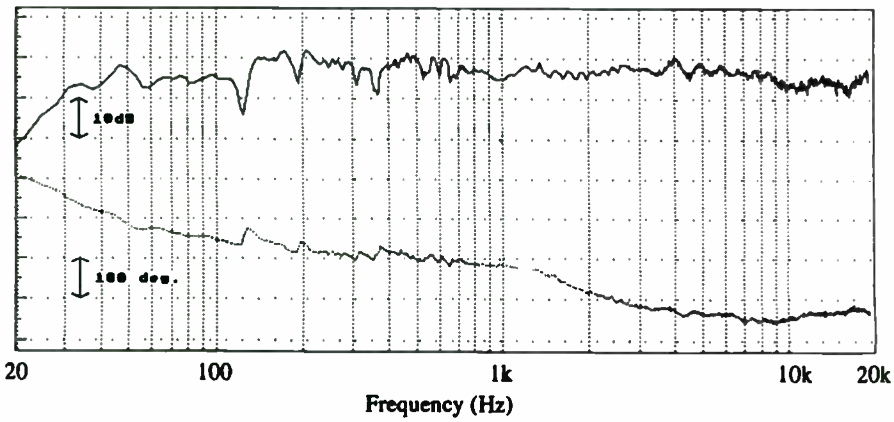

FIGURE 3: Frequency response and phase response of AX2-TD2001 combination on axis, in room, at 2m.

As previously mentioned, the horn sounded similar to the direct radiating cone, but the cepstral analysis showed that the true horn was the 150mm throat section only, with the 140° horizontal wooden flare acting only as waveguide "lips" for directivity control. The unevenness in the throat impedance was largely due to the abrupt, horn-to-lip termination at about 150mm from the driver diaphragm. The horn was consequently reclassified as a 150mm horn with 290mm lips (the shortest horn tested), explaining the similarity between this horn and the direct radiators. You should note that no absolute dividing line exists between horns and direct radiators, since a direct radiator can be considered a 180° conical horn of zero length.

PROBLEMATIC SPEAKERS

The longer horns, even those with relatively good mouth termination (usually easier to achieve in a long horn), are identified as horns by the temporal spacing of the reflections. Even when the reflections are significantly lower in level than those of the short horns, we recognize the greater separation in time of these reflections as a horn-like sound.

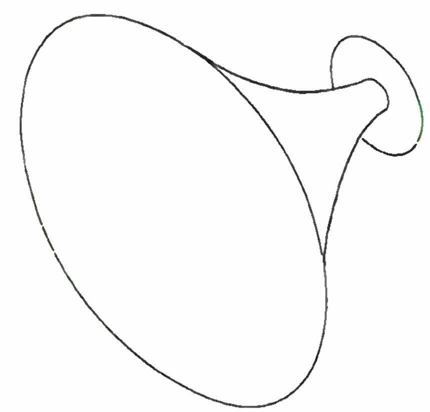

FIGURE 4: Axisymmetric horn geometry.

Cepstral analysis revealed that the two horns—one long and one short—not identified in the listening tests as sounding like archetypes *B* or *C* exhibited only minimal mouth reflections. The long horn, though showing some similarity, but not a particularly strong resemblance, to archetype *C*, was considered somewhat similar to the electrostatic (archetype *A*). The short horn showed a considerable sonic resemblance to the Tannoy dual concentric (archetype *D*).

Both the Tannoy and the Quad Electrostatic have their roots in 1950s design, yet are still in daily use in "quality control" suites. These units had their historic "difficulties" in the low- and high-frequency ends of their performance, but both had a clear midrange, suiting them well to quality control applications. Apart from reasons of inadequate (woolly) bass and limitations on maximum sound pressure level, these loudspeakers also lost favor as studio monitors because their sound was not representative of other loudspeakers in general. From the cepstral analysis the reason for this is clear, but it poses an interesting philosophical point: should a monitor loudspeaker be rejected because it does not possess the midrange problems inherent in most other production loudspeakers?

While the electrostatic (archetype *A*) was deemed similar to the sample loudspeakers on a relatively small number of occasions, it was frequently noted that one of the nine test signals (a recording of a waterfall, band-limited 1–6kHz on playback) sounded more "wet" on *A* than on any other loudspeaker; a testament to its reality.

DESIGN IMPLICATIONS

Throughout the tests, we watched for evidence of the horn's construction material showing any patterns in the sonic test results. Other than

Continued on page 34

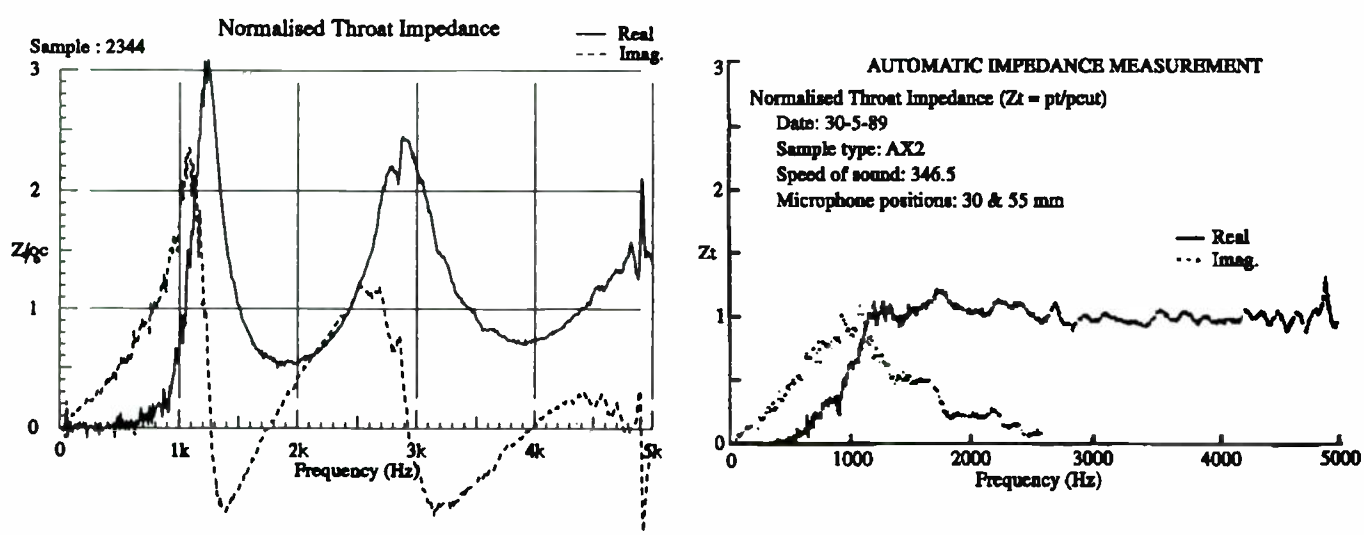

FIGURE 5: Throat impedance plot of a typical constant-directivity horn (left). Throat impedance of AX2 axisymmetric horn (right).

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Value µF	Diameter mm	Length mm	SRP US$	Value µF	Diameter mm	Length mm	SRP US$
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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
10.0	23	33	3.23	120.0	51	63	21.98
11.0	24	33	3.38	130.0	54	63	23.38

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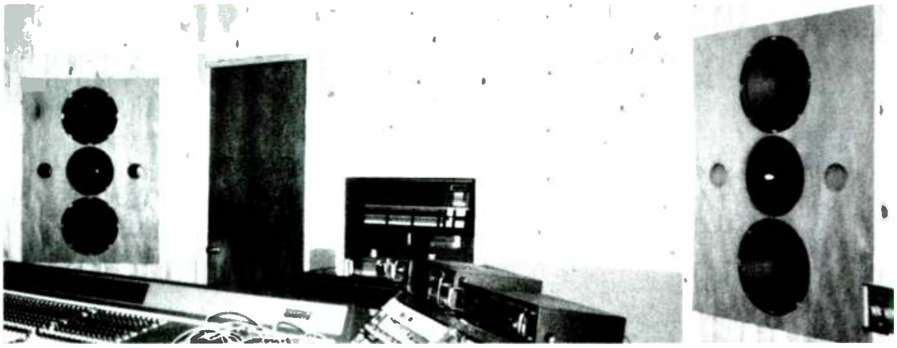

PHOTO 3: Liverpool Music House in England, where the measurements for *Fig. 3* were taken.

THE PHENOMENON OF HORN CUTOFF

The following description of horn acoustics grew out of a request to Keith Holland to answer the frequently asked question, "Why do horns have a cutoff frequency?" A thorough literature search had failed to reveal a physical explanation, and the subsequent formulation of a physical theory of horn behavior has led to useful extension of the one-parameter modeling exercise.

Horns are waveguides with a cross-sectional area which increases, steadily or otherwise, from a small throat at one end to a large mouth at the other. An acoustic wave within a horn must therefore expand as it propagates from the throat to the mouth, at a rate dependent upon the horn's local flare rate. A comparison between the wave propagation (nonexpanding and expanding) in two simple acoustic systems may help to explain the physics of horn behavior.

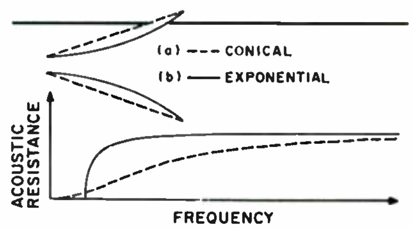

FIGURE A: Comparison between the normalized throat resistance of an exponential and a conical horn of comparable size (ignoring reflection from mouth). The acoustic resistance curves are typical of the frequency responses expected from such horns.

PLANE VS. SPHERICAL

In the first instance (nonexpanding), consider the propagation of a one-dimensional free progressive plane wave, such as a low-frequency sound in an infinite pipe. A wavefront, defined as an isophase surface, undergoes no change in cross-sectional area as it propagates, and the normalized acoustic impedance at any point along or across the pipe, ignoring losses, is purely resistive and equal to one. A plane velocity source placed anywhere along the pipe thus has no reactive acoustic loading at any frequency, either as added mass or as stiffness.

In the second instance (expanding), consider the propagation of a three-dimensional free progressive spherical wave, radiating from a monopole point source. In this case, a wavefront continually expands as it propagates, and the normalized acoustic impedance at any point is dependent upon both the distance from the source and the frequency. The impedance approaches unity, and hence is largely resistive at high frequencies and large radii, but is reactively dominated at low frequencies and small radii.

If the radius and the frequency are equal to the speed of sound divided by 2π, the impedance's reactive and resistive parts will be equal in magnitude. A spherical velocity source of finite size will therefore be subjected to either resistively or reactively dominated acoustic loading, depending on the size of the source and the frequency of vibration. The only physical difference between the propagation of waves in these two systems is the expansion or "stretching" of the spherical wave as it propagates; similar to the stretching of a balloon skin as it inflates.

WAVE EXPANSION

In the case of the plane wave, a forward or positive particle velocity is accompanied by a positive increase in pressure at the same point, due to the parallel motion of adjacent particles, and sound is thus radiated. With the spherical wave, however, because an outward positive velocity causes adjacent particles to move apart, the positive, plane wave-like radiating pressure is accompanied by a negative stretching pressure due to the expansion. The plane wave-like propagating pressure is proportional to, and in phase with, the velocity, and independent of frequency and radius. The nonpropagating stretching pressure is proportional to displacement, inversely proportional to the frequency and radius, and in phase-quadrature (90°) with the acoustic particle velocity.

Continued on page 36

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certain materials having specific problems due to bad design or construction, evidence indicated that any well-damped, solid material can be used to manufacture horns. Obviously, certain materials lend themselves more easily to the manufacture of different shapes, and some materials have probably had sonic characteristics attached to them because they are found only on certain generic designs.

WAVE SHAPES

Investigations into the wavefronts from the different horns showed that the axisymmetric designs generated waves resembling flattened spherical caps, midway between a true spherical expanding wave and a plane wave from the mouth. The waves leaving rectangular horns were spherical expanding waves which struck the walls of the horn at 90°.

Early in the tests, bubble-blowing experiments were performed: wire loops were bent into the mouth shapes of the horns. Only circular, or nearly circular, mouths produced complete bubbles; rectangular shapes caused the bubbles to tear apart before they could leave the wire.

Also, rectangular horn designs produced disturbed responses when listened to or measured from a position 90° to any discontinuity, such as waveguide plates, the top/bottom-to-sidewall junctions, and any other departures from a smooth surface. The mouth shape and any internal discontinuities tend to produce reflections from the mouth or strange aberrations in the off-axis responses. All these signs pointed to axial symmetry as the only viable option for the highest-quality reproduction.

AXIAL SYMMETRY

For public address and sound reinforcement applications, directivity is a prime factor in horn design; in studio monitoring, the on-axis ±20° response, together with an off-axis response which changes its frequency balance smoothly and uniformly, is usually far more important.

In the above tests and cepstrum plots, the axisymmetric AX2 horn was driven by an Emilar EK175 compression driver (*Fig. 2H*). A small reflection can be seen at about 50mm from the diaphragm. This was due to the slightly differing flare rates of the driver throat and the throat of the horn. When mated to the TAD TD2001 compression driver, the flares match exactly and the reflection disappears. Its falling high-frequency response also closely matches the gradually narrowing directivity of the AX2 horn, producing a smooth on-axis pressure amplitude response, together with an off-axis response where the fall-off of high frequencies takes place in a smoothly controlled manner (*Fig. 3*).

Monitor systems using the TD2001/AX2

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Vas 77.7 L
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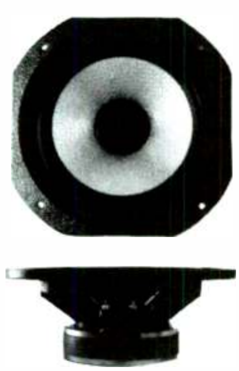

20KLV8A
8 in. woven kevlar
Cast Frame
1.5 in. VC
Welbex Anulus
80 Watts
92.1 db 1W/1M
Vas 81.7 L.
Fs 38.5 Hz.
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PHOTO 4: Regiestudios, featuring a 40-track ADTA setup, in Amadora, Portugal.

combination are now used commercially (*Photos 1–4*), particularly in very nonreflective control rooms where the on-axis response is highly important. These monitor systems, especially in inexperienced hands, suffer from some of the criticisms formerly leveled at the Quad Electrostatics and Tannoys and are not necessarily representative of other loudspeakers. But, at the same time, many experienced engineers praise the ability of these systems to pinpoint fine detail. Most studios use large and small monitor systems—one representing "truth," the other a "real world" mix—for the most viable partnership.

In many ways, the AX2 could well define the limits of midrange horn design. The axisymmetric shape (*Fig. 4*) seems to be the only one which can produce an output free of response irregularities caused by pillars, plates, other obstructions, or surface junctions. Horns much more than 300mm in length begin to produce horn-like sounds unless the mouth termination is close to perfect.

Given that the flare rate dictates the throat cutoff frequency, and mouth size controls the smoothness of the low-frequency termination to the room, then a horn with a low cutoff frequency and a smoothly flared mouth would be of such great length and mouth

Continued on page 64

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Continued from page 34

The wave expansion, and hence the stretching pressure, effectively reduces the resistive part of the impedance at low value of the distance/frequency ratio, and increases the reactive part. Such positive reactance is usually associated with added mass, but in spherical waves no extra inertia is involved when compared to plane waves. The usual description is therefore inadequate to explain this property.

In this case, the positive reactance is clearly due to "negative stiffness," not added mass. The region near a small source where this reactance dominates is the hydrodynamic near field of the source; its extent is frequently dependent. The outside region where resistance dominates is the far field.

FLARE RATES

The concept of stretching pressure can be applied to horns by considering their flare rates, which are defined as the ratio of the rate of change of wavefront area with distance, to the area of the wavefront. A conical horn has a flare rate inversely proportional to the distance from the cone apex. A spherically radiating source can be considered a special case of a conical horn, and thus shares the same expression for flare rate.

At any given frequency, the radius at which the impedance's resistive and reactive parts are equal in magnitude occurs where the above mentioned frequency/distance ratio is equal to unity. The flare rate at this radius is equal to twice the free field acoustic wave number (the wavelength divided by the speed of sound) and is identical to the flare rate in an exponential horn with this cutoff frequency. With flare rates below this value, resistive, far-field-type propagation takes place, and with flare rates above this value, reactive, near-field-type propagation occurs. Using this physical concept, the difference in behavior between different types of horns can be explained.

The flare rate's radial dependence in a conical horn, which shares the propagation properties of a spherical wave, gives rise to a gradual transition from the reactive, near-field-dominated propagation associated with the stretching pressure, to the resistive radiating, far-field-dominating propagation as any wave propagates from throat to mouth. This transition in a conical horn from near to far field, reactive to resistive dominance, is gradual with increasing frequency and/or distance from the apex; distinct "zones" of propagation are thus not clearly evident.

EXPONENTIAL VS. CONICAL

Since the property of resistive loading gives horns their greater efficiency when compared to direct radiators, as the resistive loading rises, the radiating efficiency rises. From the above, the characteristic gradual throat impedance cutoff slope of a conical horn should be readily appreciated (*Fig. A*).

On the other hand, an exponential horn has a flare rate which is constant with distance along the horn. Therefore, at frequencies below cutoff, the reactive, or near field, type of propagation dominates throughout the entire length of the horn. Below cutoff, if the horn is sufficiently long, an almost totally reactive impedance exists everywhere within the horn. Conversely, above cutoff, the resistive, far field, type of propagation dominates, again almost everywhere throughout the horn's length.

Above cutoff, propagation within an exponential horn is physically similar to a spherical wave of large radius, with minimal stretching pressure. Below cutoff, propagation is dominated by the stretching pressure and is thus similar to a spherical wave of small radius. Clearly, the abrupt cutoff phenomenon of an exponential horn occurs because the transition from resistive to reactive propagation occurs simultaneously throughout the entire length of the horn as the frequency is reduced below cutoff, producing the typical throat impedance characteristics of the exponential horn (*Fig. A*).

The use of conical horns in audio is thus limited because for practical horn sizes, the resistive loading, and hence high radiating efficiency, rises only gradually, and thus the useful high-efficiency radiation only begins to develop into a uniform response at much higher frequencies than in a comparably sized exponential horn. At frequencies below cutoff, the reactive propagation is somewhat similar to the reactive loading of direct radiators, and thus falls to the comparably low radiating efficiency of a direct radiator of the same throat size.—**PN, KH**

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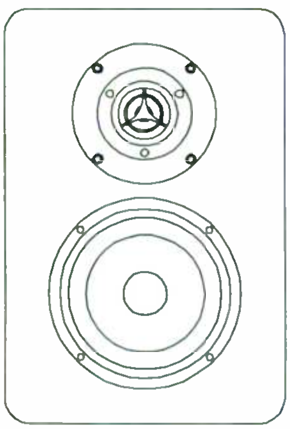

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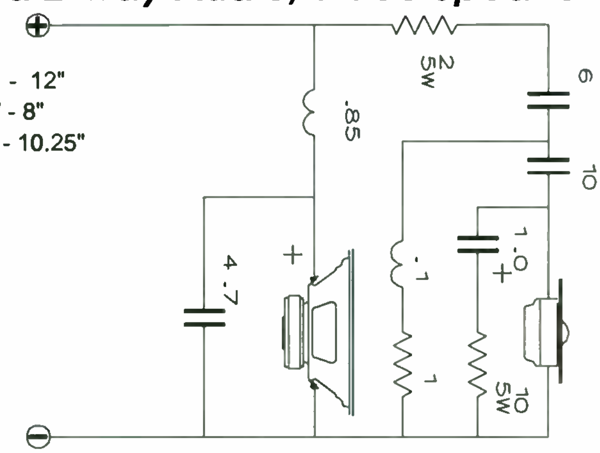
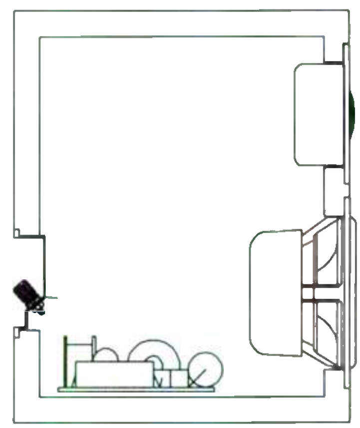
D - 10.25"





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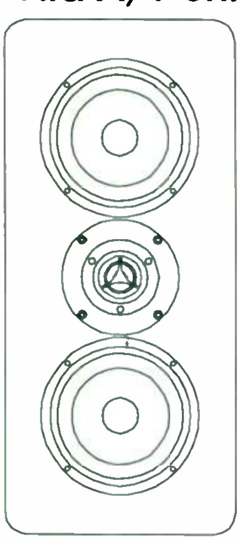

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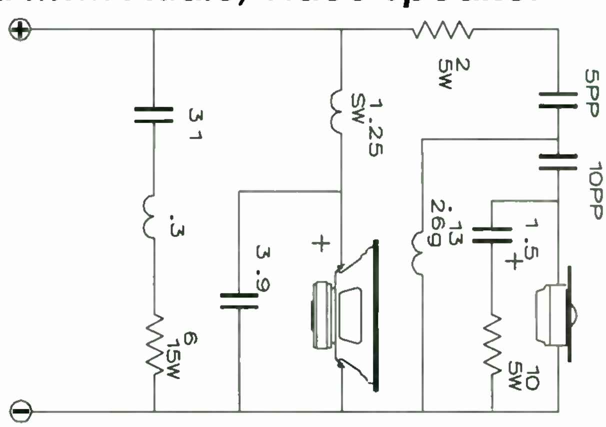
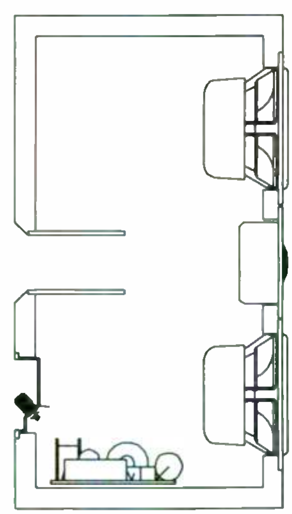
D - 10.5"





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Reader Service #8

Part 2

THE LINEAR-ARRAY CHRONICLES

By Philip Witham

If you have followed the two previous articles in *SB* on the Linear-Array concept (3/94, p. 28 and 5/94, p. 43), you know I've been building a cheap proof-of-concept speaker prototype to test whether the idea makes a large improvement in imaging. Well, I can now say that it works, and pretty much as the theory said it would, with some exceptions and unanswered questions.

All in all, Murphy let me off easily this time in the reality-meets-theory department. (We are dealing with speakers, after all, a craft known as a *strange attractor* to chaos mathematicians.) I think this was because I sacrificed 12 drivers—burned their little hearts out—and the gods were appeased.

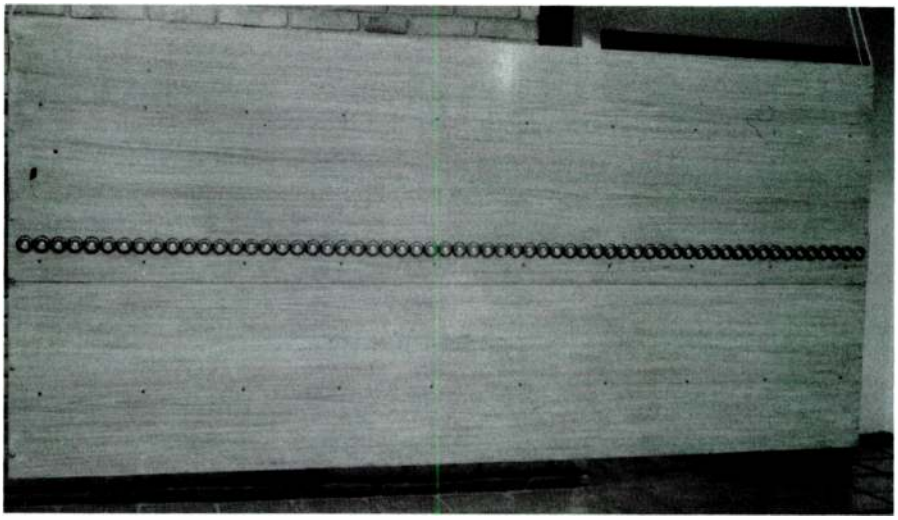

PHOTO 1: The author's 60-channel line-array speaker.

THE PROTOTYPE

I now have a 60-channel dipole speaker (*Photo 1*) using little "full-range" dynamic drivers, spaced 40mm (≈1.6") apart. The total active width is just under 8′, and the baffle is about 4′ high. The drivers are made by Kobitone (from Mouser Electronics, 25SP500).

The microphone (*Photo 2*) is built to handle 60 channels, but is only "stuffed" with 48 at the moment—6′ worth. It uses Panasonic WM-60AT omni mike capsules, also set at 40mm intervals. The mikes extend a few inches away from the horizontal structure to reduce acoustic interference. The mike can fold up to 4′ × 4′ for transport. The microphone structure is made of soldered copper tubing and fittings (plumbing parts), with a wood base.

Built on the vertical structure of the mike is the electronics box, with two 32-channel preamp PC boards (*Photo 3*). Two fat 50-pin ribbon cables plug into the microphone and speaker to handle all the interconnections (*Photo 4*). Power is from a small regulated DC supply which plugs into the back of the speaker. The DC runs along the ribbon cables to the preamp, where it is regulated again.

The little 600Ω drivers (*Photo 5*) are driven directly by the preamp's op amps, with no "power amp" stage. The total power output rating would be about 6W RMS, so we're not about to shatter any windowpanes here. *Photo 4* shows the connections at the back of the speaker. A search for 50- to 100-pin connectors and cable yielded precious few choices, most being very expensive. The ribbon cable connectors, on the other hand, are very easy to use, with little "Ejector/Latch" levers on the sockets for easy unplugging. A pair of scissors and a bench vise are all you need to make up a cable of any length, and the parts and cable are inexpensive.

Also, flat cable is available in ½" diameter round jacketed/shielded versions which are less expensive than conventional multiconductor cable. I still wound up with two cables to handle 64 channels. The mike preamp is designed to cut off the bass below 100Hz to reduce wind noise problems, and the speaker can't handle bass below this point anyhow. This limitation shouldn't get in the way of tests of imaging, though.

WHAT'S IT SOUND LIKE?

Everybody, including myself, was astonished to find that it was impossible to distinguish between my own voice, and Mr. Edison's recreation of it.—Anna Case, Metropolitan Opera soprano, 1915

A friend and fellow speaker experimenter turned me on to this little bombshell of a quote, which he keeps posted on the wall of his audio lab. So, with that as a warning, a disclaimer: I've spent enough time with the prototype to state some things as facts. But I haven't done many methodical or blind listening tests yet. So the degree to which various observations are true is still to be determined. For the sake of brevity this time, I'll avoid couching my statements in careful, scientific terms.

The following listening tests were mostly conducted with the microphone set up outdoors. The speaker was indoors, in a typical largish rectangular room. The system had only 6′ of active width and 48 channels at the time. The gain was set so that the reproduced sound level was a bit higher than the level at the mike.

Ambient outdoor noises and whatever noisemaking objects I had lying around served as the test sources. Some wind chimes hung on a stand and activated by remote control (a rope) were especially useful. Voices are also very useful. The observations of about five other listeners who heard the system only occasionally differed from mine.

THEORY vs. REALITY

When listening to the speaker, you hear the sound coming from the level of the drivers, and it is *mostly* limited to a left-to-right angle somewhat smaller than the width of the array. Certainly, the left-to-right imaging accuracy is superb. It is solid, unambiguous, and better by far than anything I have heard before. It appears equally good for distant sources (airplanes, dogs, birds) as for sounds near the mike. A sound source angle can be located when listening to the speaker, then compared to the sound outside at the mike (seen from the relative angle of the mike), and the two always match.

The imaging does not break down as you move left and right, with the apparent position of the source fixed in space until its direction puts one end of the array too close. Then the apparent direction bends around that end of the array, fixed on the end of the speaker rather than at some point behind, and the volume drops. This happens when there is less than about 1′ of extra array width to the side of the direction of the source (i.e., if you draw an imaginary line between your listening point and the virtual image point, and look at where this line passes through the speaker array, the imaging works if you have at least 1′ of extra array to either side).

This leads to reality-versus-simple-theory modification #1: I need an 8′ array width, instead of the 6′ originally planned, to reproduce a nice wide soundstage without having to sit too close to the speaker.

The image is not affected by head rotation, remaining fixed and natural. Or, I should say, it is affected by head rotation just as it is with natural sound.

The mike obviously does not distinguish between sources above, behind, in front, or below. They all sound as if they are coming from a plane on the level of the speaker drivers. This is sometimes surprising in that otherwise the imaging is so good, the complete lack of correct vertical and "surround" imaging is obvious. The overall sound is very natural, and blends into the room more realistically than stereo, but then, an airplane flying left to right in front of you rather than overhead is odd.

IN YOUR FACE

The perspective on sound sources changes with listener movement about the room, as it should. This is most startling when two sounds are present simultaneously—one a distant source, say, an airplane apparently straight ahead, and another source up close, say, chimes close to the center of the mike. As you walk from the left to the right, the airplane sound remains at a fixed angle to you. The chimes start out to your right, pass the direction of the airplane in the center, and wind up on your left. The angle between the two flips as you proceed by the chimes.

In this way, with your movement, the "depth" location of sound sources is clear. It's not so easy or clear if you stay in your seat, however. The depth sensation is much weaker than the L-to-R sense. Perhaps this is partially due to the omni microphone, which tends to pick up a lot of "ambience"—reflected sound—even with close sources. Perhaps not.

The L-to-R "soundstage" width continues to widen as you move closer to the speaker, and indeed it is almost 180° when you put your face right up there. Even at this distance most sounds do not seem to be coming from the speaker so much as "out there" (vertical height excepted). It is clear that the imaging is not just a stereophonic illusion, but honest-to-god wavefronts moving off the speaker from the different directions. There is no fixed speaker location where a sound is apparently coming from, distant sources move with you horizontally.

Sounds coming in at the mike at extreme angles ("end-fire," or nearly so) are reproduced at extreme angles. When you are up close to the speaker, they appear way down towards the end. If you back away from the speaker, they disappear, and are heard at a much quieter level "somewhere in the room." Often, from a typical listening position, these end-fire sounds are audibly louder where they bounce off the side walls of the room than directly from the speaker. Different sounds bounce about the room differently and are sometimes heard from odd directions.

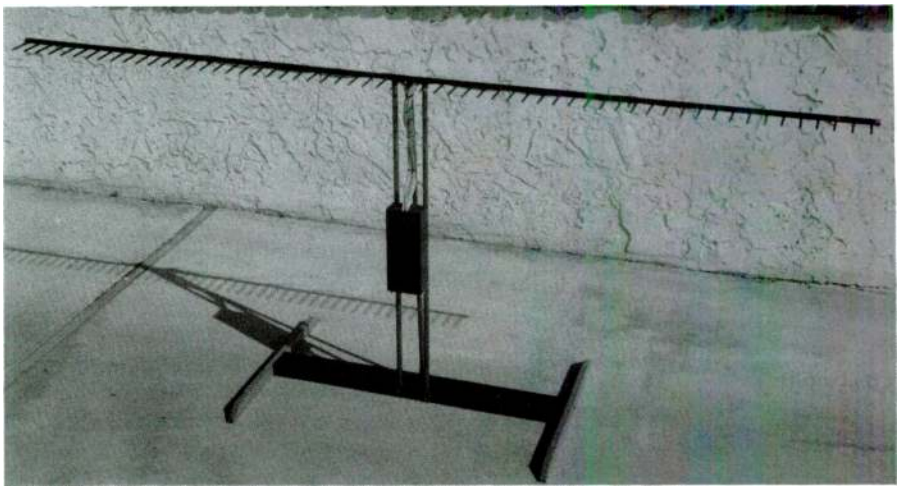

PHOTO 2: The microphone, measuring 8′ wide.

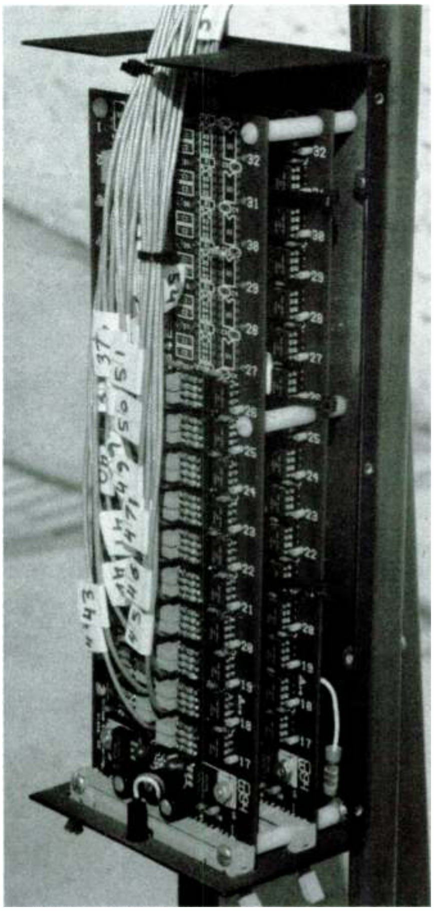

PHOTO 3: Closeup of the preamp located on the mike's vertical structure.

The dipole nature of the speaker is helping to spread ambient sound around the room behind the speaker, but I have not decided whether a dipole is the best way to go yet. It is clear that the array is producing ambient sounds which blend into the room more effectively than does stereo.

BUGS, CATS, AND CHIMES

The sound of bugs outside at night and other ambient noises are very convincing. After a while, they blend into the background as if I had several windows open, not as strong a sense of sound coming from the speaker, just an enjoyable presence in the room. Then, when I unplug the system, poof, all gone! It is rather startling to realize that subconsciously you felt it was coming in through the windows. The overall effect is stronger and more pleasing than in stereo.

I believe I need to introduce a new term to describe something important about the sound of the array. "Image complexity" would be a good term to use. Stereo by comparison sounds as though it is simplifying the sounds it reproduces. The array sound is rich in image character.

For example, my wind chimes consist of about 40 bamboo tubes and a set of ten or so solid metal rods all bumping into each other. The array reproduces this as a *bunch* of *little* sounds located in *different* places *very* close to each other (packed into a single vertical plane, unfortunately). I'm not certain how best to quantify this yet, but I will measure how close two sound sources can be before they blend together.

In comparison, stereo's treatment of this same scenario will produce a nice sound, but the images of the separate sounds will be smeared together and *simplified.* Ditto for the natural reverberation. Stereo also tends to produce a large image from small, close-miked sources. The array is inclined to make things sound more merely life-sized. I am reminded of a plot of the polar dispersion of a violin, at just one frequency, which is lumpy and varied with angle. Stereo samples this mess from a few points in space and assumes that this sample of the instrument's output is all we need to hear.

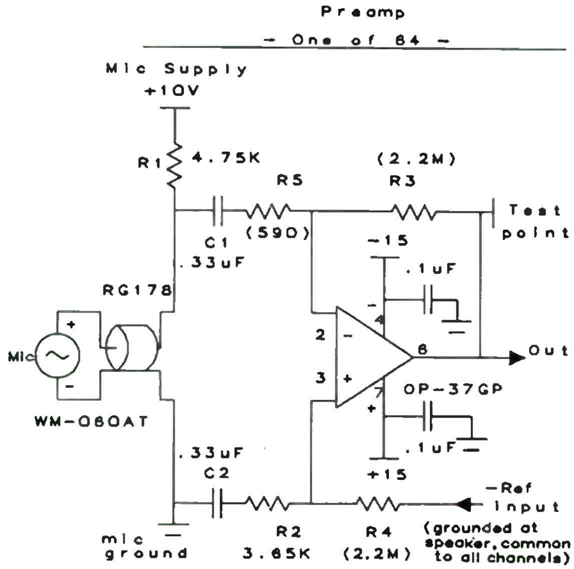

FIGURE 1: Mike preamp circuit (one of 64).

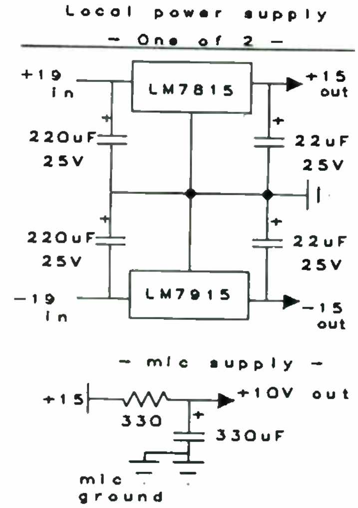

FIGURE 2: Local power supply circuits (one of two).

The Linear Array reproduces a much more complex sound. My impression is that this has a lot to do with my dissatisfaction with stereo: that the ear can take in and distinguish many separate, complex sound sources simultaneously, and cannot distinguish as many details from stereo (through speakers). I am also vaguely dissatisfied with the terms "image" and "imaging," since marketing hype and overenthusiasm have so abused these and many other related words that they lose their usefulness. I try to keep my English feet on the ground and just report what I hear.

I can attest that cats are convinced much more by the array than by stereo. I had a good laugh watching a cat's head swinging back and forth with a moving (reproduced) sound when he sat underneath the speaker, looking up. Our cats ignore the stereo system when it is playing. Incidentally, I heard the sound from the same direction his head pointed.

THE ALIAS EFFECT

As far as image aliasing and the top two octaves of the spectrum, testing with the metallic wind chimes is encouraging. The chimes produce strong, distinct 5–10kHz fundamentals according to my scope. The calculated end-fire (worst case) image aliasing limit of this array is 4.3kHz. If the chimes are placed up a few inches from the mike, most of the mike array is at close to 90° from the chimes, which is a tough test of the system. The result is an image of a number of separately located "ting" noises spread about 1′ wide where the actual spread of the chimes is 4″.

The overall sound is located where it should be, but I believe I am hearing the individual chime rods displaced up to half a foot from where they should be. I think this is aliasing, and each particular frequency is moved in its own way, like rearranging the locations of the separate rods. But this is as good as I hoped it could be. Much better than stereo!

At no time did the chimes show up *well* away from their proper locations. Some treble, which ideally would appear to come from just one area, is heard coming from all across the array. This is at a lower level than the main image, and is probably aliasing, as well as reflections in the environment around the mike.

I believe I am hearing the apparent effect of 5–15dB of extra gain effectively added to distant sources, relative to what they

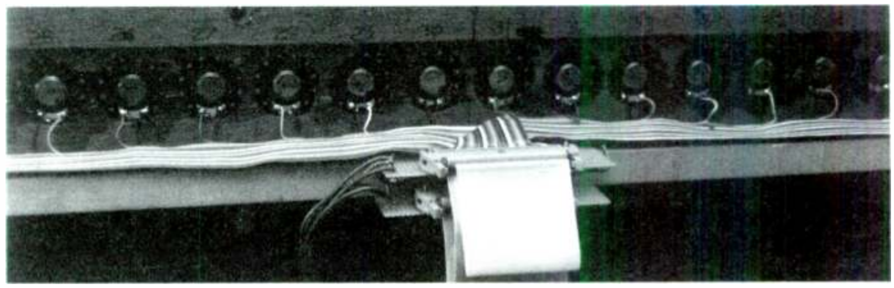

PHOTO 4: Connections and the two 50-pin ribbon cables at the back of the speaker.

PHOTO 5: The 600Ω drivers.

"should" be. Things become quieter as they move away, but not as fast, perhaps, as is natural. This effect is possibly predicted by theory: distant sources come out with narrow dispersion, while close sources are reproduced with wide dispersion. As a result, the close sounds will lose amplitude faster as a listener moves away from their virtual location. I am working to pin down this area of theory.

MORE TESTING REQUIRED

I will measure live-versus-reproduced sound levels at various distances to quantify this. But it is interesting that I can hear conversations 300 yards away just as well from the speaker as when I stand outside and listen. A related effect is the reduction in the overall mike noise level resulting from the sheer number of channels. Theory predicts that if you add more channels and reduce the gain of the individual channels to keep the same overall gain, the total noise power will be reduced to $1/N_{CHANNELS}$. Or, in terms of amplitude:

$$-10\ \mathit{Log}\ N \text{ decibels}$$

which for stereo is -3dB, and for 48 channels is -17dB. This is true as far as I can determine, since the prototype uses very noisy little 4mm diaphragm electret mikes, and I hear only a modest noise level.

The $4 "full-range" drivers have some definite problems:

1. They can't handle the full output level of the microphone preamp without breaking their super-fine voice coil wires. I've replaced 12 so far, mostly casualties of wind noise and a (ahem!) feedback experiment.

2. They have a bass emphasis around 250Hz that was not bad when I tested a single driver by itself but is bad in the full system. They attenuate the midrange from 1-4kHz and have a nasty resonance at 5kHz or so. From the back, they sound even worse.

It's surprising how pleasant and listenable this speaker is, used as a Linear Array, considering how bad it sounds when driven by mono or stereo. The bass emphasis was obnoxious, though, when a friend played his acoustic guitar for my mike. It is possible that an effect of this array configuration is a gradual decrease in sensitivity with frequency of up to 2dB per octave, overlaid on the individual driver response.

The dipole design has a few drawbacks. Sounds produced "end-fire" are also produced off the back side in the same direction as from the front, heading off towards a side wall. The two versions cancel each other well at lower frequencies. I wish to hear these end-fire sounds strongly, bounced off the walls, since this could help reproduce "hall ambiance." A dipole can't do that as well. Also, strong reflections off the wall behind the speaker occur. These may or may not help with the imaging, but I notice that the speaker baffle blocks much of it from a normal listening position.

This has an interesting effect on my stereo system, when I set its speakers up against either side of the baffle. The imaging from the stereo is much worse with the baffle there, especially in depth and height. I sense that much of the "height" imaging from stereo is the result of room acoustics. I'd like to try the array as a boxed speaker.

WHAT'S IN THAT BOX?

Figure 1 is the schematic of the mike preamp. I omit the connector pin numbers and the remote power supply. The supply produces ±19V regulated DC at up to 1A, and uses three-terminal IC voltage regulators. The power is regulated at both ends to eliminate AC hum noise coupling in the long ribbon cable, and to provide a solid supply at the preamp despite the cable resistance.

Figure 2 is the circuit of the regulators at the mike. Each preamp channel is a differential amplifier for good noise and crosstalk rejection. It is not a particularly good diff-amp in terms of common mode rejection ratio, but it is less expensive than any alternative. The IC is the lowest grade (GP) version of the OP-37, which costs about $1.80.

The circuit may be a bit odd if you are used to the standard four-resistors-and-an-op amp diff-amp configuration. R5 should be the same as R2, right? No. The microphone is used as a current source rather than a voltage source (the typical way these mike elements are used). The result is higher gain with higher amp speed (lower feedback resistor value) and lower noise. The mike's output is the drain pin of an FET, and thus works well as a current source.

The mike itself has about a 15k impedance, which in parallel with R1 becomes about 3.6k, the same as R2. C1 and C2 decouple DC from the input and, in this case, also filter out wind noise below 100Hz. R5 tunes the common mode rejection and response at high frequencies. The -Ref input to R4 is a pseudo-differential output trick that reduces crosstalk and AC hum. Due to the long cable between the mike preamp and whatever unit is at the receiving end (speaker, amp, recorder), there is likely to be a difference between the ground voltages at either end.

I am running a separate ground sense line back to the preamp from the (speaker/recorder) end of the cable. This sensed ground reference is added to the output by each preamp channel. All channels on each PC board use the same reference. Not enough wires exist in a practical cable to use a fully differential output design! The "mike supply" is RC filtered, since this point is essentially a second input to each amp that is just as sensitive as its mike input. Without this filter, the noise and crosstalk would be very high.

SYSTEM SPECS

All resistors are 1% metal film types, and C1 and C2 are polyester film (no space for more audiophile-approved types). I will parallel these with nonpolar electrolytic caps when I mike the bass below 100Hz. The mike elements and their coax cables are soldered in with no connectors, since the 0.5mA bias supply through them would play havoc with any contact resistance. I can just imagine wiggling 60 connectors to eliminate a scratchy noise!

The components with values in parentheses are built on DIP headers, so they can be swapped out to change the gain. *Table 1* lists some values for various gains. I've only tested the preamp at 60dB gain, so be forewarned if you use this circuit that the lower gain settings may require changes. The op amp may oscillate at high frequency at lower gain. A 100Ω resistor in series with the output to the cable should solve that. You may need to substitute an OP-27 (the OP-37's slower sister) or another type for a 20dB gain setting.

The gain numbers I list are relative to the voltage you would see from the mike into the normally used 2.2k supply/load resistor, or across the mike's internal 2.2k resistor. Note that the very high gain I am using is intended for driving the speaker directly, and is so high that the preamp will clip with a rather low sound level at the mike. A more appropriate level for use with an amp or recorder would be 20–30dB.

The total electronics noise to 20kHz, measured with a resistor in place of the mike, is 2.1μV RMS (referred to the input). I can't

TABLE 1

GAIN SETTING VALUES

Gain	R3,R4	R5
60dB	2.2M	590
50dB	698k	187
40dB	221k	59
30dB	69.8k	18.7
20dB	22.1k	5.9

measure or hear any AC hum from it. The noise from the microphone element measures 70µV RMS. Obviously, the preamp doesn't add much to the total noise level. The maximum output level is 9V RMS. THD is high with the 600Ω speaker load and 60dB gain.

At 8V output level, THD is typically 0.01% at 200Hz, 0.1% at 2kHz, and 1% at 20kHz. Without the speaker load, THD is 0.05% at 20kHz, and about 0.005% at 2kHz. THD would also drop in proportion to the gain setting. The frequency response of the preamp is +0,-0.1dB from 800Hz to 21kHz, and -3dB at 70kHz. The low cutoff is -3dB at 132Hz. Crosstalk at the speaker terminals between any combination of channels is -50dB or better at any frequency, which is better than you need.

FUTURE EXPERIMENTS

Before formal testing, I will fully stuff the system out to 60 channels and eight glorious feet. Then I will adjust each individual channel for equal gain. The mike sensitivities currently vary by ±3dB or so. Then I will conduct the following tests in as unbiased, methodical, and blind a manner as practical, with a handful of individual listeners:

- Closed box around back of speaker without the large baffle—differences versus dipole?
- Source location in contrast with apparent position map
- General performance of 30 channels (80mm spacing)
- Minimum separation between sound sources before they blend together, 30 and 60 channels
- General listener comments—preferably to live music
- Frequency response at various source and listening positions (1/3-octave warble tones, calibrated to remove mike and source errors)
- Volume level versus source and listening distance, live versus reproduced

WHAT'S NEXT?

After all that, we'll have definite proof of the merits of the idea, and I'm sure the next steps will be to build a better speaker and more electronics, and to beg or borrow a gang of ADAT recorders. I could then record music and cart the system around for demonstrations.

I'd like to clear up a bit of terminology that seems to be misused: a line array with enough channels for the job is *not* "N-channel *stereo,*" since it does not rely on your hearing mistakenly fusing the separate speaker sound sources to produce a "stereo illusion," but actually reproduces the various different wavefront directions regardless of whether a human is listening or not. And no Zen jokes please! (If no one is there to listen, does your stereo image well?...aargh.) It *is* an "N-channel line array." The hearing mechanism in use is your natural binaural one.

A correction to the last Linear Array article in the 5/94 issue: the polar plots were captioned as "Measurements," which they are not. They are simulations. Lastly, I'd like to thank Mr. Dell and *Speaker Builder* magazine for their support and interest.

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This new loudspeaker engineering tool for Windows combines visual interface and electroacoustics into a flexible loudspeaker designer and analyzer. Modeling, small signal analysis, and design for ported (standard or isobarik), sealed (standard or isobarik), bandpass, tubed, and horn-loaded enclosures are provided. For ported enclosures, the design can be changed to see the effects of different port frequencies and/or box volumes. Bandpass enclosures of orders four through seven can be designed and effects of changing the rear and/or front box tuning, or changing the rear and/or front volume, can be tested. Tubed enclosures, ideal designs for truck speakers, can be designed for any diameter. Large signal analysis is provided for ported (standard, series isobarik, or parallel isobarik) and sealed (same) enclosures. An active high pass filter (orders one through four) can be used to act as a subsonic or rumble filter. Large signal analysis shows the relative on-axis response (SPL) at one meter, as well as the maximum input wattage per frequency. Port dimension calculation for all designs and one to four ports is also available, as is enclosure dimension calculation for all types covered.

LADP also can design high- and low-pass filters of order one through four. Individual types include Linkwitz-Riley, Chebyshev, Legendre-Papoulis, Butterworth, and Bessel. Bandpass filters of order one or two can also be constructed for all the above types except Bessel. Small signal analysis can show up to three filters at a time and include the dB response and phase angle. Further, the polarity of the high-pass filters can be reversed to show the effects of in-phase and out-of-phase filter designs.

Loudspeaker directional quality can be seen graphically as a polar plot. The off-axis response is shown from -90 degrees to +90 degrees. This can be a useful tool in determining the upper cutoff frequency of a low-pass or bandpass filter, and the high frequency "beaming" effects of the speaker can also be tested. If you have a sinewave generator, voltmeter, frequency counter, and 1000-Ohm and 10-Ohm resistors, you can also measure Thiele-Small parameters. A graph of the voice coil impedance in free air can be shown, as well as the output parameters: Vas, Qts, Qes, Qms, Ras, Mas, and Cas.

Some basic architectural acoustics such as room reverberation, impulse response, and room modal frequency distribution can also be explored with LADP. A graphical room modal frequency response will help the user to determine how the room reacts to certain frequencies, as will the decay response to a tone burst for frequencies at 125, 250, 500, 1000, 2000, and 4000Hz.

On-line help provides a supplement of information for each dialog box. It also contains helpful hints for designing, building, and testing enclosures and crossovers, as well as general car stereo information. Background information on acoustics, electrical/stereo components, and speaker designs is also available, as are a glossary, program overview, program utilization info, graph interpretation guide, and definition of every prompted input.

NEW IN LATEST VERSION: addition of network drive search at installation time; replacement of vented enclosures with eq by sixth-order active-filtered enclosures, schematic and print capability; replacement of sealed enclosures with eq by fourth-order active-filtered enclosures, schematic and print capability; addition of cone excursion and port air exit velocity calculations to the Loudspeaker Sound Pressure option; addition of summed loudspeaker(s)' acoustical response and loudspeaker enclosure system phase modeling into the Crossover/Driver Interaction option; improvement of bandpass enclosure modeling transfer functions, with more custom designing options for bandpass analysis; improvement of Windows system resource usage in the repaint routines; and addition of passive radiator design and analysis.

All output can be printed to the Windows default printer in either portrait or landscape mode. System requirements: 2Mb RAM, 3.3Mb hard disk space, MSDO5.0 or later, Microsoft Windows 3.1 or later, math coprocessor. 2 x 3-1/2" DS/HD.

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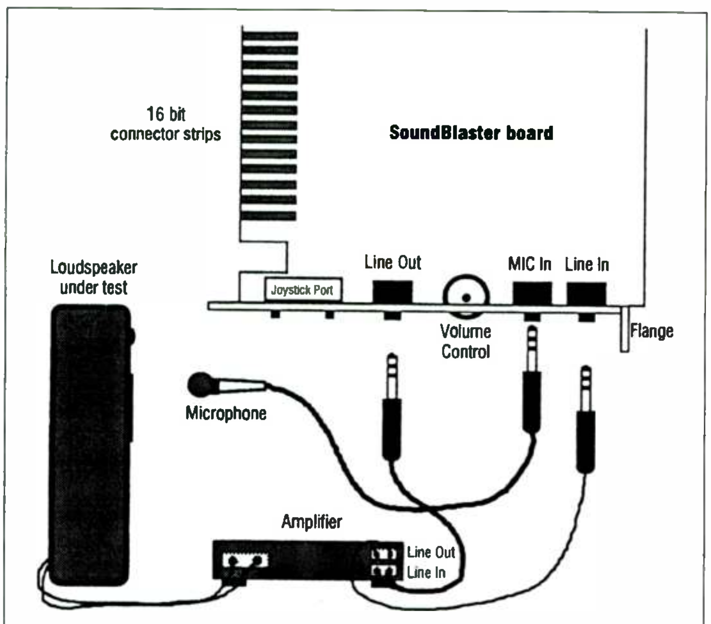
AIRR (ANECHOIC IN ROOM RESPONSE) SOUNDBLASTER SOUND CARD SPEAKER RESPONSE SOFTWARE — SOF-AIR1B3G — $49.95

Julian J. Bunn

NEW FROM SWITZERLAND! AIRR is a program that offers a cheap alternative to loudspeaker measurement for owners of PCs equipped with a SoundBlaster or compatible sound card. Based on the principle of pulse mode excitation of the loudspeaker followed by Fast Fourier Analysis, AIRR features full color real-time plotting of both the digitized pulse received by the microphone and the frequency response of the loudspeaker system. AIRR is controlled by a single key stroke and data entry at the keyboard, and full on- line help is supported. You can toggle between several modes of operation, which include Loudspeaker/Amplifier response measurement, Time/Frequency averaging, Square/Sin(x)/x pulse types, and Pulse On/Off. For anechoic measurements, cut markers allow a window to be placed over the time data in order to remove the part due to room reflections. Frequency response spectra obtained can be stored as ASCII files on disk, which then can be subsequently read back in by AIRR and subtracted from and/or overlaid on the current spectrum. Using the subtraction feature allows corrections to be applied for amplifier or microphone response. The AIRR display itself can be "captured" into a PC Paintbrush (PCX) file, which may then be included as graphics in a document.

The AIRR software includes operating instructions combined with on-line help, last minute changes and instructions, the AIRR article itself from *Speaker Builder* 8/94 (in Word), and a sample file containing microphone response data. AIRR requires a PC equipped with a SoundBlaster 16 sound card (or compatible, such as the MediVision Pro 16); VGA screen with at least 640 x 480 resolution, preferably color; microphone (usually supplied with the sound card, although a calibrated microphone such as Old Colony's Mitey Mike is often desired); and loudspeaker to test! Math coprocessor not required. 1 x 3-1/2" DS/DD.



THE COMPLETE GUIDE TO HIGH-END AUDIO — BKAP1 — $29.95

Robert Harley

Written for beginners and experienced listeners alike, this is the first book that tells you everything you need to know about buying and enjoying high-quality hi-fi. In 13 easy-to-read and fact-filled chapters, the author, a Consulting Technical Editor for *Stereophile*, shares secrets on how to get the maximum enjoyment out of audio. This book has it all, from nuts and bolts tips on hi-fi equipment to suggestions on becoming a better listener.

With this book, you'll find out how to get high-end sound at low prices; identify the weak links in your present system and upgrade where it will do the most good; discover how to tweak your system to get maximum performance from equipment you already own; learn how to read equipment reviews and determine why the components that sounded so good in the showroom don't sound so good at home; identify the qualities that will make you a more appreciative and perceptive listener; and best of all, learn how to keep hi-fi fun! 1994, 450pp., 7 x 10, softbound.

CHRISTMAS IN EARLY AMERICA CD: 18TH CENTURY CAROLS AND ANTHEMS — CDAL1 — $16.95

The Columbus Consort

This unique CD will make a great stocking stuffer for any audiophile, including you! Recorded in The Netherlands and produced in Germany by Channel Classics, *Christmas in Early America* brings a new sound to the holiday season, as well as an interesting and superbly crafted conversation piece. The Columbus Consort consists of the delightful mix of two violins, cello, bass, organ, and vocal quartet. Thirty-page booklet in English/French/German includes historical background as well as lyrics. 1993, 55:40.

Contents: [1] "How Beauteous Are Their Feet" (Supply Belcher). [2] "How Beauteous Are Their Feet" (Samuel Holyoke). [3] "Anthem for Christmas" (Benjamin Carr). [4] "If Angels Sung a Savior's Birth" (Joseph Stevenson). [5] "Th'Almighty Spake, and Gabriel sped" (Samuel Holyoke). [6] "Comfort Ye My People" (Samuel Holyoke). [7] "Hark! The Glad Sound" (John Jenkins Husband). [8] "While Shepherds Watched Their Flocks by Night" (Musgrave Heighington). [9] "While Shepherds Watched Their Flocks by Night" (Jacob French). [10] "While Shepherds Watched Their Flocks by Night" (William Billings). [11] "Unto Us A Child Is Born" (Johann Friedrich Peter). [12] "Prince of Peace, Immanuel" (John Antes). [13] "Hail Infant Newborn" (David Moritz Michael). [14] "To Us A Child Is Born" (Johannes Herbst). [15] "Thou Child Divine" (Johann A.P. Schulz). [16] "Meine Seele erhebet den Herrn" (Jeremiah Dencke). [17] "Hosanna! Blessed He That Comes in the Name of the Lord" (Christian George). [18] "Magnificat anima mea Dominum" (Charles Theodore Pachelbel).

SOUND SYSTEMS FOR YOUR AUTOMOBILE — BKS48 — $16.95

Alvis J. Evans, Eric J. Evans

Whether you're starting from scratch or upgrading, this book will show you in easy-to-follow steps how to plan your system, choose components and speakers, and install and interconnect them to achieve the best sound quality possible. Installations on specific types of vehicles are discussed, including separate chapters on coupes and sedans, hatchbacks, pick-up trucks, sport utility vehicles, and vans, so you're sure to find the information you need for your vehicle. Also included are a complete glossary and an entire chapter devoted to speakers. 1993, 124pp., 6 x 9, softbound.

AMAZING KOONCE UTILITY DISK #1 — SOF-KUD1BXG — $7.95

G.R. Koonce

This valuable disk contains two programs to benefit just about anyone! CONVERT1.EXE displays instantaneous, simultaneous English and metric values for linear, area, and volume measurements in inches, feet, yards, millimeters, centimeters, and meters. The volume screen also includes liters. On-screen help (hardly needed) is available. PRECRES2.EXE requires VGA to display its amazing wares: instantaneous interpretation of metal film resistor color bands! You can either enter a value and see the band colors appear on the screen, or enter the colors of your mystery resistor and have the program tell you its value. Good for up to 10 megohms, and includes three significant digits, multiplier, and tolerance. Printer option. IBM, requires VGA, 1 x DS/DD—PLEASE SPECIFY DISK SIZE.

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Wayland's Wood World

GRILLE FRAMES

By Bob Wayland

The treatment you give your speaker grille attracts attention to your craftsmanship and shows off the care and pride you have put into your enclosure design. Sure, the real part of your speaker is everything else, but we are judged by that first appearance. This column is for the builder who wishes to finish the speaker enclosure with an attractive window dressing.

TABLE SAW JIG

Most enclosures are square or rectangular, although some interesting ones are not. The traditional approach sets off the exposed speakers with a frame, onto which a covering or grille—usually of cloth, metal, or plastic—is placed. I will first describe a table saw jig that helps ensure the grille frame pieces are cut to the correct angle, which I call the *reference angle*. It is the angle made by the frame at any corner of your speaker. If your speaker is rectangular, the reference angle is 90°. (I will also suggest how to use this guide with radial arm saws.)

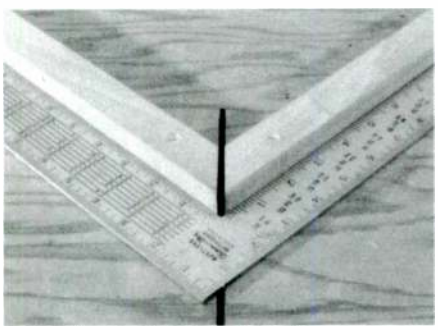

PHOTO 1: Using a framing square to check the reference angle on the jig.

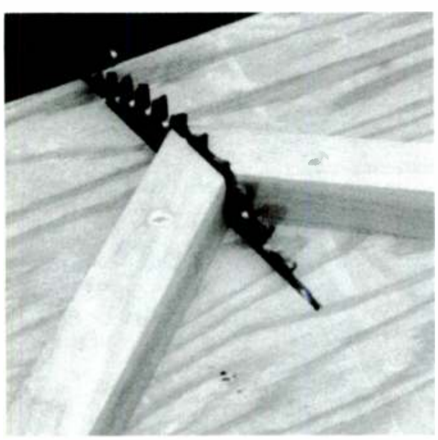

PHOTO 2: Cutting the slot in the jig. Note that the top of the saw blade has just passed the apex of the guide.

You can easily adapt the design for angles other than 90°; just use the non-90° angle for the reference angle. If your enclosure is irregular, you can have more than one reference angle. For the smallest size jig you need a *mounting board* of at least 16″ × 24″. I used an old piece of ¾″ A/B pine plywood, but just about any flat surface will work fine.

Now you need to cut two *guide bars* to fit snugly into the miter slots on your table saw. These should be made from hardwood or metal. For most saws, the cross-section dimensions are about 3/8″ × 3/4″. The length of the guide bars should be at least the width of the mounting board (16″ if you are using the minimum size jig). You will gain a bit of stability if you cut the guides a few inches longer than the minimum. The problem, however, is that the darn things stick out, so you must be careful not to catch yourself on the jig or other protrusions.

Place the guide bars in the miter guide slots on your saw. Carefully align the mounting board so that the long dimension is perpendicular to the miter guide slots and the centerline of the mounting board passes through the saw blade. Drill guide holes through the mounting board into the guide bars, being careful not to drill into your saw table. Remove the mounting board and drill out the guide holes so they are about 1/16″ larger than the flat-headed screw you will use to attach the mounting board to the guide boards.

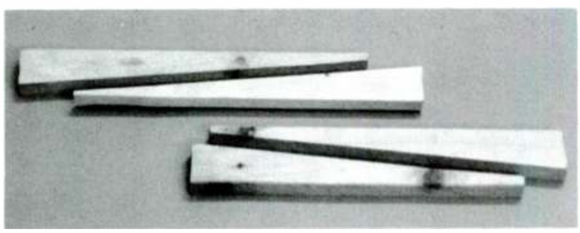

PHOTO 3: Wedges for clamping the sides in place on the jig. It is a good idea to cut a few extras.

Be sure to countersink the screw holes. If you are using wooden guide bars, put some glue on them; otherwise, drill and tap the guide holes in the metal bars for the attachment screws. With the guide bars in their slots, replace the mounting board on the guide bars (making sure the long dimension is perpendicular to the miter guide slots). The enlarged hole will allow you to make any fine adjustments that you might need.

CUTTING SQUARE CORNERS

Next you will need a guide, mounted on the mounting board, for your reference angle. If you plan to cut a number of different angles, make a set of separate guides for each. The easiest reference angle guide to make is a solid isosceles (two sides equal) triangular piece of wood with the base opposite the reference angle. Of course, it is critical that your saw is perfectly aligned when you make the guide. Attach the guide to the top plane of the mounting board so that the side opposite the reference angle, the base, is parallel to the long dimension of the mounting board (perpendicular to the guide bars). The apex of the

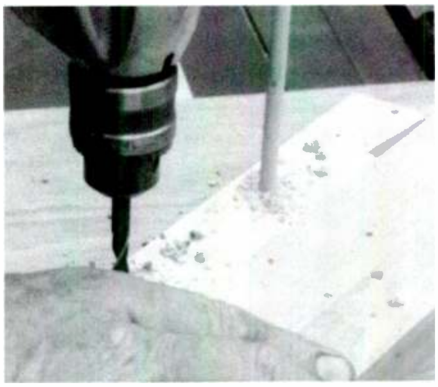

PHOTO 4: Aligning the wedges to give proper clamping of the frame sides.

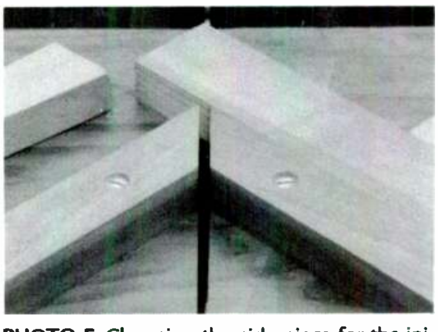

PHOTO 5: Clamping the side piece for the initial cut on the jig.

triangle should be about 3" from the front edge of the mounting board and on the centerline of this board.

You can accomplish the same effect as the reference angle guide by cutting two guide strips, at least 12" long, from straight 1" × 2" stock. Place the strips on the mounting board so that they intersect the centerline at an angle one-half the reference angle. *Photo 1* shows how to do this for a reference angle of 90°. (It is important that the strips exactly form the reference angle.) Attach the strips to the mounting board at the correct angle about 3" from the forward edge of the mounting board to the intersection of the guide strip with the board's centerline. If you will always be cutting the same angles, it is a good idea to glue the strips in place; however, if you plan to cut different angles, attach them with a minimum of two screws each.

CLAMPING DOWN

Once you have attached the reference angle guide(s), put a fine-toothed blade on the saw you will use to cut the miters for the corners of your grille frame. Then, with the jig in place, cut just past the apex of the guide(s) over the arbor of the saw (*Photo 2*).

When you use this jig, clamp the sides of the grille frame to be cut tight against the reference angle guides. If you don't clamp the sides, they will almost always slip during the

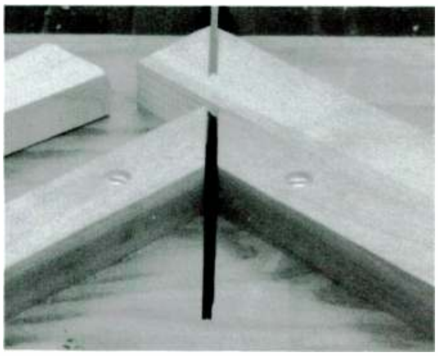

PHOTO 6: Making the initial cut of the clamped side piece.

PHOTO 7: Removing a tight wedge.

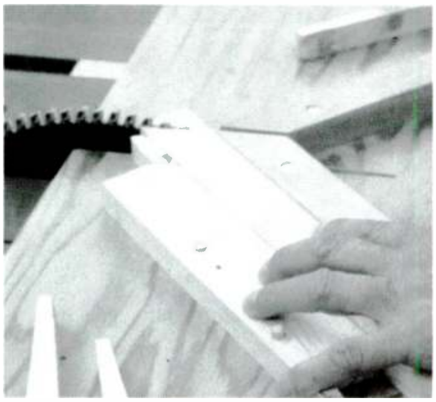

PHOTO 8: Cutting the opposite end of the side piece. Be sure to check that the piece has the correct orientation.

cutting and produce an unwanted gap in the corner joint. You can do this with C clamps or with hold-down clamps that attach to the mounting board. When you use C clamps, you risk denting the wood and having the clamps get in the way of the saw blade. But the hold-down clamps are relatively expensive, and you'll be spending money for something you will only occasionally use.

The wedge is a simple, inexpensive alternative. Cut four wedges (*Photo 3*) from scrap stock, the same thickness as the grille sides, about 10" long and 1.25–1.5" wide at the thickest part. On the long dimension of two wedges measured in thirds, drill ¼" holes perpendicular to the flat side at the first and second points. Now with the grille side piece next to the reference angle guide, place one drilled wedge and one undrilled wedge as shown in *Photo 4*.

Using the holes as guides, drill a ¼" hole into the mounting board. Put a ¼" dowel into the hole to act as a guide, and drill the other hole in the wedge so that the top edge is parallel to the side piece. Repeat this operation on the other side. Cut four 1" long pieces from a ¼" dowel and insert them through the drilled wedges and into the mounting board. Now it is a simple matter to tap the wedges tight to clamp the sides firmly in place.

Also, if the side pieces are smaller or larger than it is possible to clamp with your wedge spacing, drill a new set of holes in the mounting board and move wedges to fit as needed. Or you can drill a series of holes with about 1" spacing and be ready for anything!

You can also use this same basic jig with a radial arm saw. Make the jig as described above, but don't attach the miter guide bars and don't cut through the mounting board. Clamp the mounting board to the saw table so the saw blade will cut through the apex of the reference angle guide. It is essential to accurately align the kerf with the saw blade each time you use it.

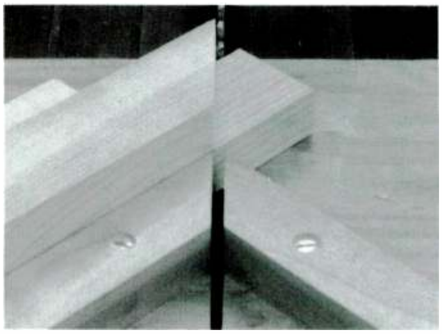

PHOTO 9: Using a cut piece to check the alignment for making a second identical piece.

CUTTING A GRILLE FRAME

The wood you use to make your grille frame should be straight-grained and free of knots or other irregularities. If you have a choice, use quartersawn boards, often sold by specialty lumberyards for facings on cabinets. To be absolutely certain the frame members are exactly the same thickness and width, you can often cut all the frame members from the same board.

After choosing the material for the grille

Continued on page 66

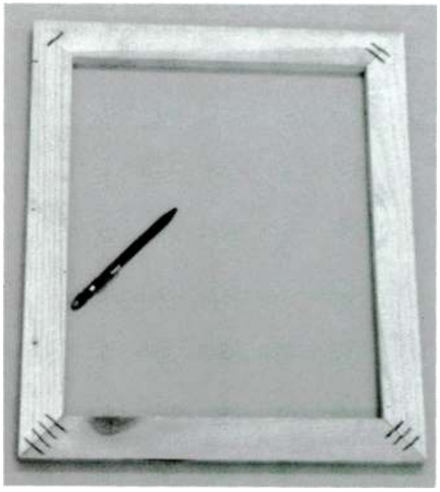

PHOTO 10: Marking the order of the joints for your frame. In practice, you need only make light marks that can be easily erased, not the heavy ones used here for illustration.

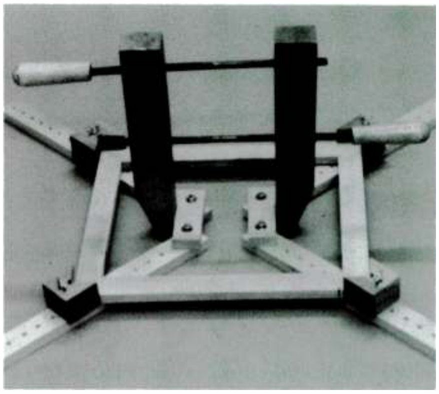

PHOTO 11: Clamping the grille frame for gluing.

Software Report

PC AUDIOLAB

Reviewed by Richard Campbell
Contributing Editor

PC AudioLab©, Microacoustics Audio Software Products, 2553 Carpenter St., Thousand Oaks, CA 91362, (805) 495-8945. System requirements: IBM 386/486 AT; 4MB RAM memory; VGA graphics display; SoundBlaster© or Media Vision© 16-bit sound card.

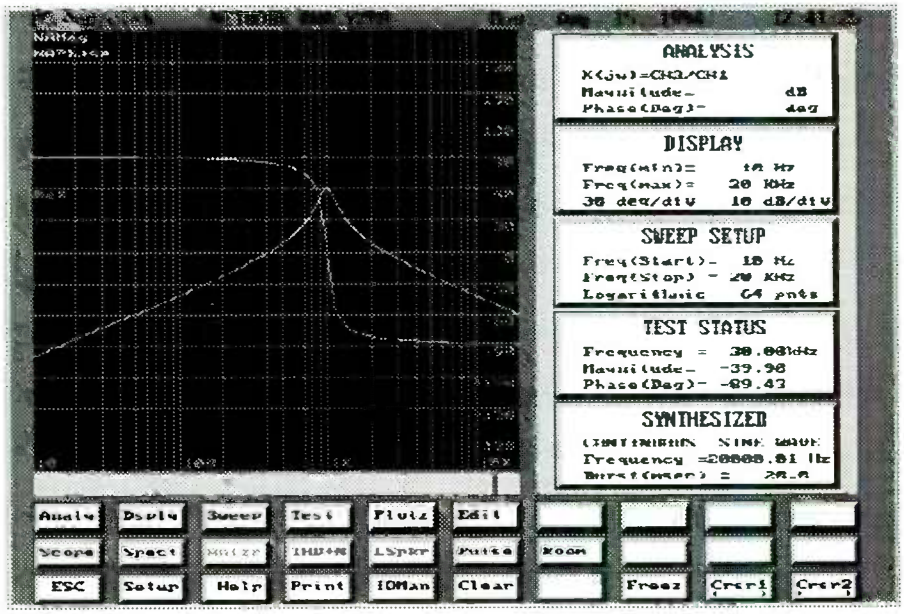

FIGURE 1: Typical PC AudioLab display screen

PC AudioLab from Microacoustics Audio Software Products is a very ambitious program. Since I did not run all of it under ideal circumstances, I will start this review with a few words of counsel.

A review for *SB* readers is difficult at best for any graphics-intensive software package. Everyone reacts differently when faced with the personal nature of the user I/O (i.e., what you must do to the mouse or the keyboard to get some action, and what you see on the screen as the result). Each user is apt to love or hate the things that, respectively, I hate or love. It's all about what we are used to.

This program turns a PC into a digital laboratory instrument, of which I have used many, so my interpretation of its actions is judged in light of this experience. Another person currently using an oscillator, voltmeter, and scope would be mighty impressed by its capabilities. My annoyance might be your deliverance.

A program's full capability is best learned from the manufacturer's brochure. If you like what I say, send for the brochure and decide on the functionality and applicability to your case based upon that information and not wholly upon this review. I have six different loudspeaker/box design software packages which give exactly the same numbers because they can each add two and two and get four, but they differ dramatically in how they relate to me, the user.

Software should be robust, stable, nimble, unintimidating, well structured, thoughtful in screen presentations, and come with a very good manual. After several days of working with PC AudioLab I am beginning to have fun with it. I have mastered the keystrokes and mouse buttons, and I am inside the programmer's head. Since the software drives a commercial multimedia I/O card, nimbleness is *not* one of its virtues.

WHAT'S INSIDE?

Your PC essentially becomes a two-channel digital scope with lots of bells and whistles. The program author supplied me with a beta test Version 2.0, and I used the SoundBlaster 16 board from Creative Labs as the hardware interface for signal generation and data acquisition. This popular card is widely used in MIDI and multimedia applications, so many potential PC AudioLab users need only purchase the software. The program addresses the sound card directly, not requiring any SoundBlaster software drivers or environment statements. It can also talk to the Media Vision Proaudio Spectrum 16 card.

An important fact is that the package uses a DOS extender, in this case one called Ergo. It manages memory above 1MB in its own way, and is a good solution for memory-hungry programs and for those which use DMA and need large blocks of RAM with data structures habitually transcending 64K segment boundaries.

The big problem with DOS extenders is that they conflict with existing memory managers. In fact, conflicts are so certain no matter how much RAM you have that you might just as well trash all the memory managers in your CONFIG.SYS file before attempting to run this program. After all the blood, sweat, and tears working to stuff everything you need into

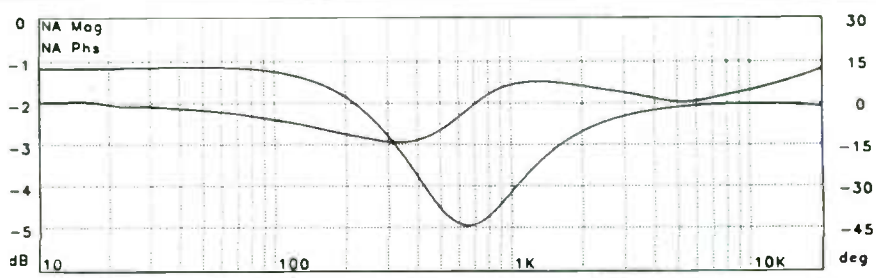

FIGURE 2: Transfer function magnitude and phase for a passive notch filter with a 5Ω resistive load.

RAM's nooks and crannies, you can run this large and complex program with just two statements in CONFIG.SYS: FILES=40, BUFFERS=30.

I suggest making a clean-boot floppy, or activating one of the modern DOS configuration selection features that can show a menu prior to boot. You can call a menu item "audio," which will then produce an ultra-simple boot for running PC AudioLab. *[MS DOS 6 and Novell DOS 7 both allow a multi-boot facility to configure memory in various configurations.—Ed.]*

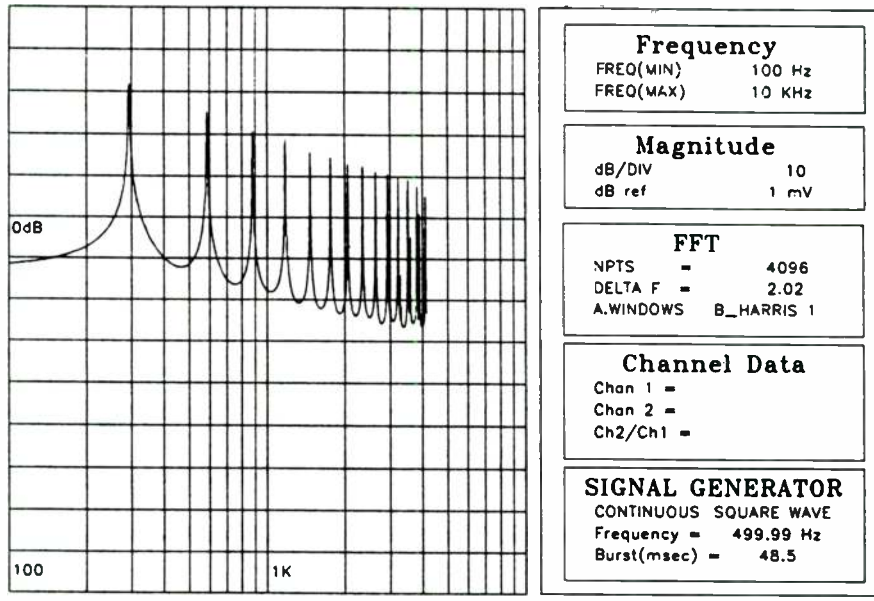

FIGURE 3: Spectrum analysis of 50% duty cycle half-triangle function generator output.

INSTALLATION

The entire program installs into a single directory consuming 2,964,240 bytes. My beta copy uses the SoundBlaster card, so the distributed version may take more or less space. Also, the disk space allocation is dynamic, as the program always saves the last thing you were working on and restores it next time around. A software security system requires the master floppy be in a drive to start the program. Installation takes only a few minutes, and is effortless thanks to the EZI© installer. If you don't prepare your machine for a clean boot prior to installation, there are plenty of dire warnings about what to do before running it.

I had clean-booted my computer but left the disk cache functional. Even though I have a 486-33 with eight megs, the presence of SUPERPCK caused a bizarre problem alluded to in the manual. Repeated punching of the left mouse button caused the screen to scroll up. An error message apparently was being written in the same color as the background, and when it got to the bottom the background scrolled up. Goodbye, SUPERPCK. I'm now stuck with slow disk reads, because when I call for a screen dump the graphics print driver takes forever.

Problems with the software protection scheme were largely my fault. Whenever I tried a new function I had to rearrange my lab, so the program ran in fits and starts. Seems like every third time I tried to boot some bizarre event would happen. More than once I started it with the master floppy in Drive B to see the Drive A light come on, then go out—*novissima verba!* Once I was interrupted by a phone call, and when I got back Drive B was spinning and a message read "put master floppy in drive __"—*obiit sine prole!* Several times the message "cannot enable A20" appeared, and if I simply retyped the command line everything was fine. (Maybe I should consult Wotan.)

I can't blame the program author for incorporating protection, but other truly stable systems are available. Furthermore, they allow backing/restoring the protected directory to floppies or tape, provided the directory is restored on the same computer. I run a suite of room acoustics analysis programs with a retail value of $11,000 which are software-protected in this fashion. I am allowed a preset number of loads but an unlimited number of restores from a backup copy onto the same CPU.

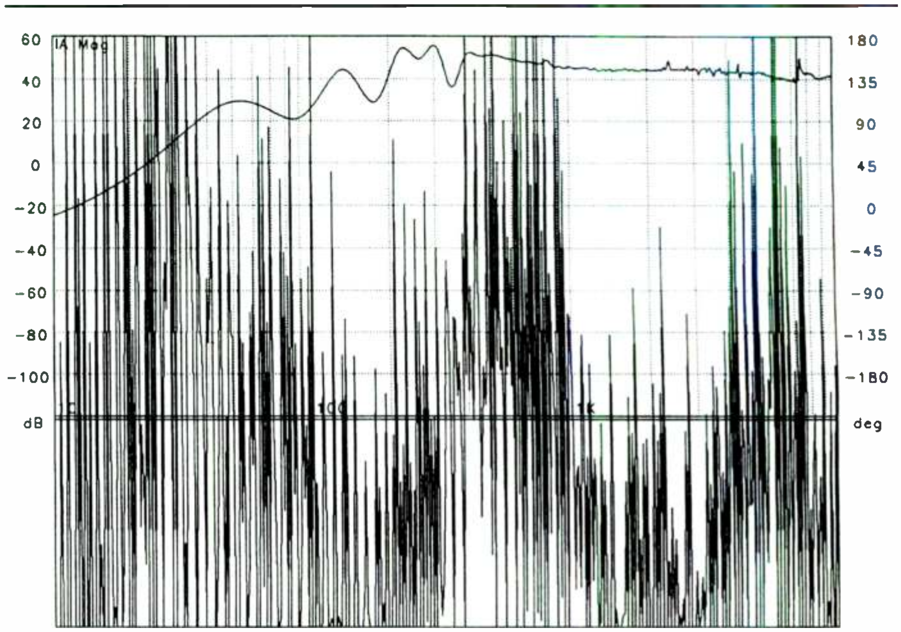

FIGURE 4: Loudspeaker impulse response 2cm from cone using a GRSLM calibrated microphone. The bottom window is supposed to display the average of ten impulses, but the scaling is incorrect so the points are scattered vertically over the whole plot.

GETTING FAMILIAR

You work with screen pushbuttons. As you move the mouse, different buttons flash and the left mouse button (LMB) activates a particular one. The scope display is on the screen's upper left, the acquisition and signal settings are reported in five large boxes to the right, while the action buttons are along the bottom. I can't show you a typical screen from my computer because, of the three screen grabbers I own, none would allow PC AudioLab to load with them while they were TSRs. Plenty of screen shots appear in the manual, however, so something must work with this program. The program author has kindly supplied a screen shot (*Fig. 1*).

The three rows of ten action buttons are used depending upon which measuring function is currently active. The bottom row controls the system and readout cursor; the middle row selects a function; the top row allows you to detail that function. In the event your mouse gets eaten by the cat (which the manual says sometimes happens), there are logical ways to access the screen buttons: F1-F10 for the top, Alt-same for the middle, and Ctrl-same for the bottom. Unfortunately, no clues appear on the screen for these substitutes.

As a first step, be sure the software can talk to the SoundBlaster card. Push the <Setup> button and read what the top right box reports for DMA channel and I/O address. Between the SoundBlaster and PC AudioLab manuals, plenty of good advice is available for getting it right. A <Help> button elicits some highly condensed but well-written contextual help inside the upper left scope-screen box. I used DMA Channel 5 and port 220 for the SoundBlaster.

SCOPE & SIGNAL GENERATOR

Now let's see if the signal generator will run. Plug a Walkman-type headset into the SoundBlaster output port, BUT DO NOT PUT IT ON BECAUSE YOU DON'T KNOW WHAT THE AUDIO LEVEL WILL BE! Place the headset where you can hear it squeal. If <Oscil> (scope) in the middle row does not light red, push that button first. Then select <Sigen>, which appears on the top line because it's a scope detail setting. Ahhh...the mouse won't go to <Sigen>.

Once the scope is switched on, and while it is reading and processing the input to the SoundBlaster card, it cannot service the mouse. Two solutions to this problem exist.

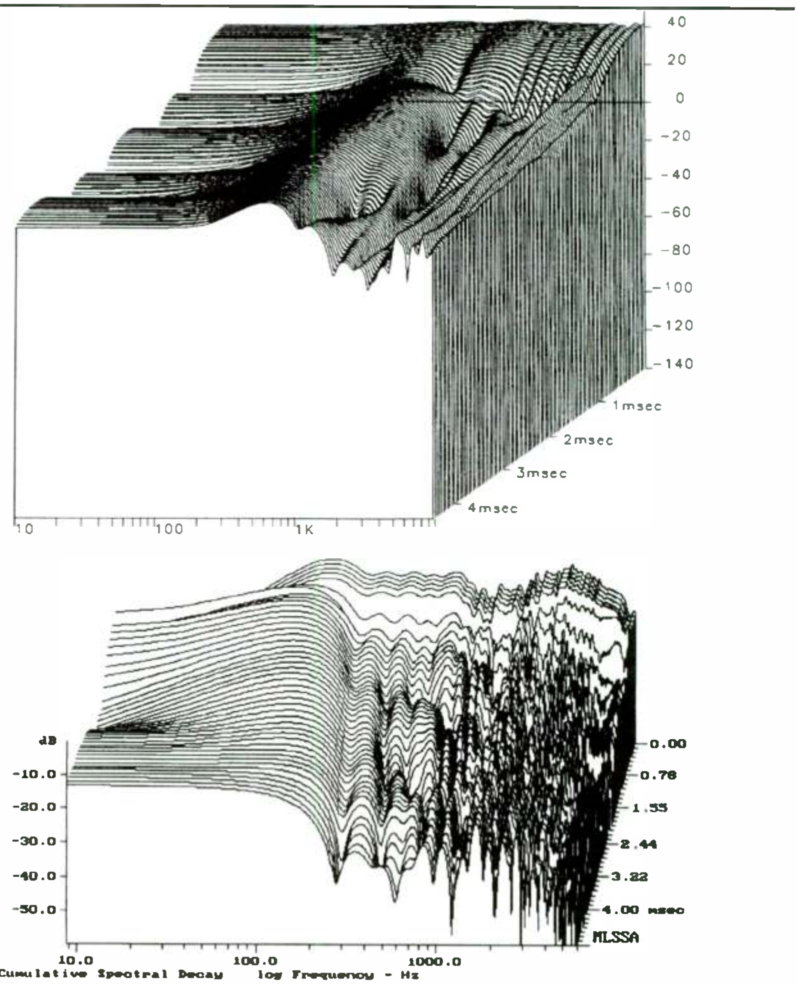

FIGURE 5: Cumulative spectral decay plots from PC AudioLab (top) and MLSSA (bottom) for the same experimental setup. The top plot illustrates the need for additional facility for scaling and zeroing the data.

One is to type Ctrl-F8, which is a <Freeze> button. At the end of the acquisition cycle, this button turns red and you can now use the mouse to freely access the other buttons. The second solution is to hold down the LMB until the acquisition cycle stops, placing you

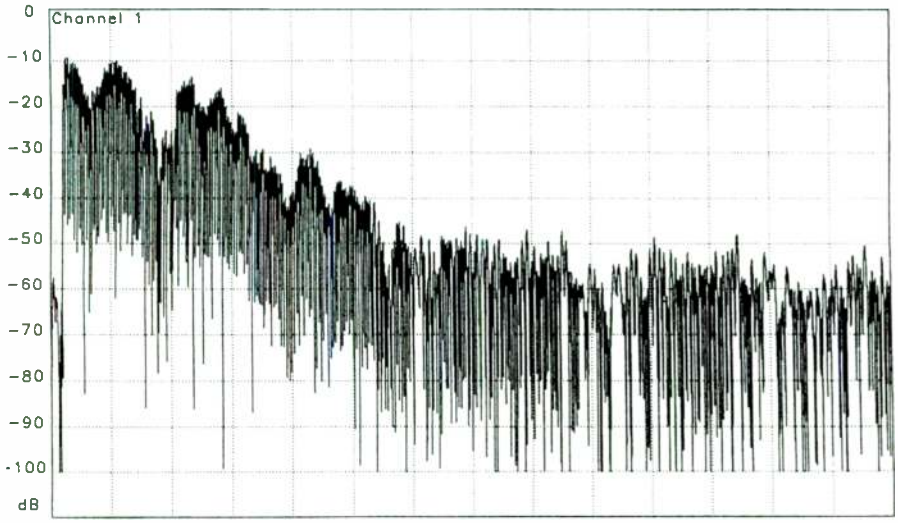

FIGURE 6: Room analyzer plot of small lab space using 20 averages of a 500Hz tone burst.

in a temporary freeze. This allows you to move the mouse to the desired button and activate it by releasing the LMB.

Clicking on <Sigen> changes the top row of buttons to <Style>, <Freq>, <Wave>, two-burst control, and <Done>. Click <Style> to continuous, then select <Freq> and type in 1000.00 Hz, and <Wave> until sine appears. Click <Done> and adjust the SoundBlaster volume control to hear the tone.

You now have a sense of how PC AudioLab looks and feels. As with all programs, it takes a while, but the rewards are really worth it because this one can do lots of things.

DISTORTION ANALYZER

With this great feature you can select the start and stop frequencies of a log or linear sweep between 10Hz and 10kHz, and the number of test points up to 1,000. Forty log points provide a suitably smooth screen display.

The program dwells at each frequency long enough to collect data and compute an FFT, then selects the second and third harmonic amplitude relative to the fundamental for plotting. The dwell is about two seconds per point, so you have a full frequency display of distortion in a minute and a half. Very useful. The residual distortion (output connected directly to input) climbs from 0.1% at 100Hz to 1% at 10kHz.

Just for fun, I ran the sweep through the RA2100 reference power amplifier and connected its output to a resistor in series with a 1A, 40V Schottky diode to ground. I then made a distortion run on the resulting nearly half-sine pulses and the harmonic levels were dead-on.

NETWORK ANALYZER

This very flexible function allows you to take ratios of both channels' FFTs, for example, to compute transfer function in magnitude and phase. As a test, I chose an all-too-familiar loudspeaker passive equalizer on which I am working, with the EQ's output connected to Channel 1. I ran the test signal from the card directly into Channel 2, told the program to trigger on that, then plotted the ratio of Channel 2 to Channel 1 (*Fig. 2*).

I have data from MLSSA and also computed in SPICE for this network, and the PC AudioLab data was essentially identical to these methods. For other purposes, you can choose from several combinations of sums and ratios.

LOUDSPEAKER ANALYZER

Two features of this analyzer include an acoustic and an impedance measurement. The manual says SPL, which is fuzzy because the term implies a measurement system whose 0dB point is calibrated to 20μPa RMS sound pressure. This equipment can measure only relative level in decibels. I noticed several files in the directory concerning microphone calibration, but no mention is made of this in the manual and no such screen button is displayed. I suspect that my manual has not caught up with my software version. Let's assume there will be a provision for mike calibration in the V2.0 release.

A provision also exists to measure acoustic phase response. The program asks you to type in the distance from loudspeaker to microphone, although I'm not sure what it does with that information. If the intention is to compensate for the fixed propagation delay, it is virtually guaranteed never to work properly. You simply do not know which part of the driver (or the cabinet) is responsible for the radiation at any particular frequency, especially as the wavelength gets smaller.

One of MLSSA's more elaborate sections measures acoustic phase by the only sensible method I know: locate the exact acoustic center for a defined frequency band by flattening the excess phase response and crank in that value of delay time. The PC AudioLab acoustic phase response zigzags all over the screen and is an utterly useless plot. Of course, in flaming yellow it's the brightest thing on the screen. At least you can click it off if you don't like it.

The distance you enter is possibly used in some sort of TDS analysis, but I doubt it. Of course, the relative SPL plot is quite useful.

The impedance measurement does what it is supposed to do, except that the display can't be scaled (impedance phase can be scaled 2×). The display window has 0–40Ω for the lower two-thirds of the left axis. When I measured a 4Ω speaker system with an R_{ES} of 15Ω, the plot was all scrunched at the bottom. This particular measurement has the potential to destroy the sound card input circuits if you become brainless about return signal limitations. By switching on cursor readout, you can get the exact value (to experimental accuracy) anywhere along the curve.

SPECTRUM ANALYZER

Figure 3 is a printout of the spectrum analyzer connected to an external function generator set for 50% duty cycle half-sawtooth at 300Hz. The beautiful distribution of harmonics is a clue as to the reason this peculiar waveshape appears on most high-quality function generators.

The spectrum analyzer control panel has two buttons, <Burst> and <Click>, which allow triggering of either a programmable tone burst or an impulse from the sound card output port while PC AudioLab is in analyzer mode. Additional comments about some of the parameter choices for the spectrum analyzer are included later in this review.

IMPULSE ANALYZER

Clicking the <Pulse> button produced a FORTRAN run-time error as it tried to read a configuration file. After comparing it with other *.CFG files and noting the possible corruption, I "rolled my own" and it worked. The resulting impulse configuration was pretty weird, but once into the program I modified it. I suspect that something went wrong in the file decompression process during installation.

Many people who could run PC AudioLab with little trouble would be stopped in their tracks by that kind of error. The manufacturer's FAX and phone lines would be buzzing. That's why, upon opening a new software package, you look first for the page in the manual defining customer support. If you don't see a mailing address and a telephone or FAX number, send it back!

The impulse scope screen is divided into two sections: the top looks like a scope; the bottom shows the actual impulse response coming into Channel 2. The concept is that Channel 1 receives the impulse directly from the signal generator and Channel 2 receives the response from the device under test. Both channels undergo a 2,048 FFT, and the complex ratio is taken CH2/CH1. The resulting magnitude and phase are shown in the upper portion of the display box.

GLITCHES

While acquiring impulse responses from a test setup, the Channel 2 input to the SoundBlaster card went dead. I recycled everything but could not get it to respond. The card had to be replaced, which caused a major delay in completing this review. The timing was terrible, because 48 eager acoustics students had just begun their eight-week course, effectively depleting my brain for that period.

Much later, with a new card, I was able to resume my work with the impulse response section and compare it with MLSSA measurements. I positioned a precision SLM microphone in the near field (2cm) of a small box-mounted driver and ran ten consecutive accumulating pulses. I connected the sound card's output to an HP465A DC-coupled power amplifier set to a gain of ten and the mike preamp output directly to the sound card, knowing the SLM would not overload the input (however, if an overload does occur, PC AudioLab beeps a warning). Everything looked fine on the screen. You can watch each time domain pulse in the bottom window and the accumulating frequency domain FFT in the top window.

Then something went wrong. The upper window was fine, but the lower one blew up into a scattering of dots all over the screen. It even printed that way, as you can see in *Fig. 4*. I suspect that the sum of the N-point amplitudes was not being divided by N to make the final plot.

A peculiar facility available in this mode is the ability to edit the frequency domain plot, granted on the grounds that sound cards occasionally produce display-ruining glitches. You correct them with the mouse—actually move the amplitude points around to wherever you wish.

Figure 5 shows the cumulative spectral decay waterfalls produced with PC AudioLab and MLSSA using the same test setup. I tried to rescale the former, but changing the front-end acquisition gain didn't work. (There are still seven empty buttons—can we have one that moves the plot up and down?)

The plot can be rotated in 90° increments for viewing all sides of the box. If you see one you wish to print, however, you're in for a surprise. View #1 is always printed no matter which one is on the screen.

ROOM ANALYZER

I ran this function only to verify that it works. I do a lot of room acoustics measurements, and I can't compare this program's simple results to the factors I consider important. *Figure 6* shows a typical result of applying this feature in a small noisy room: an echogram of 20 averaged 500Hz tone-burst acquisitions. It looks like 40dB of dynamic range if you eyeball the envelope.

A lot of data post-processing is required to elicit useful room acoustics quantities which are comparable to current mainstream calculations. The cursor readout seems to move only one index at a time, even with the middle (high-speed) mouse button pressed. With this much data to scroll, some 10× or 100× jump facility is needed.

PRINTING

No driver is available for an ink-jet printer, so I selected HP LaserJet. Fortunately, HP DeskJets talk PCL and do very well with laserjet drivers, but I had to set the resolution to 150 DPI to get a dark enough printout. Since ink jets are so popular, I hope they will be added in a future release. Either the printer drivers take a long time to load or the page image is being assembled on disk, because there is much churning before the printer comes alive.

THE MANUAL

A lot of reasonably well-presented information is contained in some 80 pages: clear screen shots, well-written text (if a bit terse), and an extensive Q&A-style review at the end encapsulating frequently asked questions (FAQs to us info highway nerds).

If you have no background in sampled-data theory, you will be scratching your head a bit. Admittedly, an applied software manual is not the place to provide such information, but it would be nice if the author had provided a few references.

For example, in the chapter on the Spectrum Analyzer, the section describing the effects of full- and half-Hamming, Hanning, and Blackman tapering windows (selectable within the program) takes three paragraphs. In one of my colleague's textbooks, 19 pages are devoted to this same subject, preceded by 533 pages of preliminary material. Some well-meaning speaker builder may be unaware that something went terribly wrong in his analysis setup.

CONCLUSIONS

PC AudioLab is a viable contender for those who have only $300 to spend on a PC-based digital acquisition, processing, and display program. I believe its problems will be sorted out in time. And, if you need help running it, you will talk with a very knowledgeable and helpful person who intends to continuously maintain and upgrade the program.

Although using a multimedia I/O card for this purpose is great in theory, it has real problems in practice. The usable output signal is 100–500mV. There must be a whopping output coupling capacitor on the card I was using, because even with a 25k resistor to ground it took many seconds to discharge the floating 2.5V DC. You could be unlucky enough to connect the PC card output to a system that does not want any DC. Both of my test power amplifiers are direct-coupled, for example, and if I connect them to the sound board output port I could get an unexpected and substantial pulse into the load.

To use the program you must invest in auxiliary equipment. A low-distortion, wideband voltage amplifier with precision gain steps from 1 to 20 would be useful in the signal source path. And a correctly applied precision voltage divider is essential to protect the PC card's input circuits from excessive return signal. You'll need two of them. In addition, a pair of back-to-back diodes across each input would help. In a printed warning message Microacoustics disclaims any damage to the sound board. *Caveat emptor* or *caveat venditor*—you must decide.

Even though PC AudioLab emulates a scope, I wouldn't run it without a real one, as well. I'm wary of a signal's actual condition or waveshape to trust the sound board out of hand. Maybe with enough experience....

PRAISES & COMPLAINTS

Most Mind-Blowing Feature: The cursor control buttons are ingenious, but it would take too many words to explain how they work. If the author actually created this technique of button/mouse control, I sure hope he patented it.

Most Annoying Features: (A) It's too s-l-o-w. I am biased because I use high-speed digital instruments every day. MLSSA displays a 65,536 FFT-based incoherency plot (full-frequency, full-excitation distortion) in the time it takes PC AudioLab to get through its first data point. Of course, MLSSA is ten times more expensive. (B) An input channel overload must continuously fire interrupts, because it's devilishly slow moving the mouse to the button that provides relief.

Most Worrisome Feature: The odd run-time error that pops up for no apparent reason. Is there a universal divide-by-zero or file read-error trap yet to be written? Also, what about the message "cannot enable A20" which occurs once every five or six start attempts? Why only occasionally on the same computer with the same boot sequence? Digital systems aren't supposed to work this way!

Most Frequented Feature: <Ctrl-F8>. It's easy to get messed up pressing screen buttons while the acquisition is in progress, even though the manual repeatedly says BE PATIENT. <Ctrl-F8> eventually lights the <Freeze> button, and you breathe a sigh of relief knowing you can manipulate through button fields without delay and without screwing something up.

Most Intelligent Feature: The screen layout is really elegant and makes you feel as though you are using a real instrument (which it is).

Cutest Feature: The little red light that tells you when the acquisition is in progress, although I've not yet figured out exactly how it relates to the acquisition cycle.

Most Comprehensive Features: Signal generator waveshape selections; network analyzer plotting choices.

Potentially Most Useful Feature: The cursor readout, but only if a mouse button sequence or keyboard command could automatically locate the cursor on successive peaks of the FFT display. Too much fidgeting is required to find the peak, and the VGA screen display resolution doesn't help. Also, why can't the cursors read out from the auxiliary channels? Why does the cursor read 33.65dB at 495.62Hz, and the exported value for the same data point reads 35.48dB?

Feature I Personally Dislike: The severe limitations imposed by the I/O hardware. (Nothing to do with the program, of course.) You must have external auxiliary amps and voltage dividers to apply the system to most audio situations. I also noticed significant leakage of the sampling frequency from the SoundBlaster card during a portion of the acquisition cycle which could raise havoc with other laboratory instruments.

Dumbest Feature: Measuring acoustic phase based upon typing in the distance from driver to microphone.

Features I Would Like to See: Automatic protection against Nyquist sampling violations; one-keystroke move of cursor(s) to FFT peaks; average-max-min readout of certain data over a test band.

More Advice: Do not export zero-frequency point on FFTs; show number of PSACC averages on screen as they accumulate; get rid of acoustic phase (or do it differently); exported data should be quote and comma delimited.

In the event you are scratching your head about some of the digital signal processing words and phrases used in this review, I would like to recommend a book which is available through Old Colony Sound Lab, *Digital Signal Processing Experiments* by Kamas and Lee [*#BKPH7, $34.95 plus $3 shipping in USA—Ed.*]. This is a great DSP beginner's book with software and a unique connect-the-boxes structure which is easy to learn. Happy sampling!

MANUFACTURER'S RESPONSE

I'd like to thank Dick Campbell for his extensive and thoughtful review of our software package. In the lengthy process of his review, PC AudioLab started from Version 1.0 and progressed to a beta test copy of Version 2.0. The advantage of sending a beta version to a reviewer is that they get to see the latest/greatest version; the disadvantage is that they review the "warts" before they can be removed. Subsequently, many of the glitches and headaches in the beta version which Dick experienced have been removed (including the plotting malfunction in Fig. 4, *the "odd run-time errors," and so on). As anyone who is familiar with software knows, nothing is truly stable and "bug-free" (I can't even count how many times my Windows programs have crashed, destroyed files, and the like). For his problems with the disk protection scheme and intermittent "cannot enable A20" error message, I would recommend finding a good computer exorcist, since the vast majority of other users haven't shared his experiences and for those few who did, there was a simple work-around.*

A few comments about the LoudSpeaker Analyzer acoustic phase measurement are in order. When the SPL measurement is first started, the user is given the option of entering the distance of the microphone to the speaker to compensate for a fixed propagation delay. Rather than making assumptions about a loudspeaker's minimum phase behavior and using Hilbert Transforms, PC AudioLab will actually measure the phase shift. If the distance entered is zero, then the resulting phase plot, while accurate, will indeed "zig-zag all over the screen," since the measured phase includes not only the drivers' inherent phase shift but also the phase shift introduced by the time delay between loudspeaker and microphone. There are several ways to determine the correct distance so that a minimum phase plot is generated. Using either the Impulse Analyzer or Room Analyzer, one can determine the flight delay of an impulse or tone burst.

Another important reason for entering the loudspeaker-to-microphone distance is when you wish to combine a near-field swept frequency response with the far-field impulse response. In order to avoid a discontinuity in the combined phase response at the merge frequency, both the near-field and far-field measurements must be referenced to the same acoustic plane. This is accomplished by applying a time shift to remove the appropriate delay from each measurement. An excellent article by Struck and Temme (JAES, June 1994, Vol. 42, No. 6) describes this method of obtaining the complete free-field response of a loudspeaker without the need for an anechoic chamber. By using the LoudSpeaker Analyzer for the near-field response and the Impulse Analyzer for the far-field response, we can merge these two test results to arrive at a full-frequency loudspeaker response.

A few comments about loudspeaker impedance measurements. Both the SoundBlaster and Media Vision sound cards have small on-board amplifiers (4W output). We recommend using this amplified signal output for impedance measurements since the line-out jacks of these sound cards can't drive the low impedance loads of most loudspeaker drivers. The impedance measurements should be performed using low excitation voltages (<1V) since Thiele-Small parameters are based on small-signal models. Using this test setup will minimize the chance of overloading the sound board inputs. The line-out jack of these sound cards should be used for other measurements, especially THD measurements, since the on-board amplifier adds approximately 10dB to the noise floor (which is why the reviewer residual distortion measurements were higher than normal for these cards).

We've taken several of the author's suggestions to heart so that you can now change the impedance plot scaling to make it easier to display low-impedance drivers and measure Thiele-Small parameters. The Spectrum Analyzer can now perform FFT smoothing along with power spectral averaging (PSACC) and will also display a 1/3-octave bar graph RTA display. The Room Analyzer section has been modified so that the envelope of response is calculated and displayed along with improved cursor movement. Also, if microphone calibration data is available, PC AudioLab will use this data for measurement compensation.

PC AudioLab was developed for audio professionals and amateurs who may not have the necessary $20,000+ to purchase the best laboratory gear or can't take all of that test equipment to a remote location. We're flattered by the comparison with such equipment, and, as shown in this review article, the results obtained by our program (using a $120 sound card) are identical. Soon we will support Creative Labs latest SoundBlaster board, the AWE32. With this board, maximum length sequence (MLS) analysis will also be possible. We're not aware of any other test measurement system that can do so much for so little.

Robert D. Watson
Microacoustics

EMULATING DIPOLAR AND BIPOLAR SPEAKERS

Loudspeakers come in all shapes, sizes, finishes, price ranges, and sound radiation patterns. Let's focus on the latter group, specifically concurrent front and rear radiation models, which may be either dipolar or bipolar.

In dipolar systems, the back radiation is a mirror image of the front, but occurring 180° out-of-phase with the front. This situation happens most often with planar speakers, such as electrostatic or planar magnetic, in which the front and back waves are simultaneously generated by the back and forth movement of a diaphragm. Thus a positive pulse to the front is accompanied by a rarefaction at the back. At the lower frequencies where the sound wave length is longer than the width of the driver, it is nearly omnidirectional, wrapping around the speaker and causing some degree of cancellation between the out-of-phase front and back waves.

With bipolar systems, the back radiation is in phase with the front radiation, eliminating most of the front-to-back cancellation. Bipolar speakers have another set of drivers facing the rear and wired in-phase with the front-facing drivers.

Compounding these effects is the reflected sound from the wall behind the speakers. Proper positioning of the speakers can be very critical in obtaining the most benefit from this reflected radiation. It can add greatly to the quality of the music, but can also cause frequency response aberrations.

BACKWARD THINKING

There has always been a faction of music lovers who prefer the imaging and added ambience of sound reproduction possible with di- or bipolar radiation. Over the years, many box-type loudspeaker enclosures have used one or more rear-firing drivers, usually a tweeter and midrange, to add ambience from rear wall sound reflections.

At least one current loudspeaker is fully bipolar, with an equal complement of drivers front and back, wired in series. Then there is, of course, the 25+ year-old Bose 901 series that has eight of its nine drivers facing the wall behind the speakers. Most of the dipolar and bipolar speakers on the market are expensive, and many have the reputation of requiring tedious experimentation with placement to derive the best sound.

If you're considering dipolar or bipolar speakers, but don't wish to invest large sums, some reasonably priced enclosures designed primarily for use as side speakers in a surround sound system are now available. The downside is that they may have sonic problems of their own when used as the main speakers in a system. They may not have been designed with complete spectral accuracy in mind, given their intended role as ambience enhancers.

If you've tried to emulate dipolar or bipolar speakers by placing similar or identical speakers in close proximity, with a forward- and a backward-facing pair, you may have experienced unsatisfactory results, irrespective of their dipolar or bipolar wiring configurations *(Fig. 1)*. My experiments have shown that the problem with this paired arrangement occurs when the rear radiation sound level is equal to the front; there will often be only a very limited location range in which they will sound good. The offset centerlines of the front- and rear-firing drivers possibly contribute to the placement problems.

SIMPLE SOLUTION

My variation, with which I am quite happy, uses a 25Ω variable resistor from Radio Shack in series with the "hot" terminals of the rear-facing speakers *(Fig. 2)*. This resistor decreases the output of the rear speakers about 10dB relative to the front when in the fully decreased (25Ω) position. My setup is dipolar in effect because of the out-of-phase wiring of the front- and rear-firing speakers, but could just as easily be bipolar. The dipolar configuration sounds better to my ears.

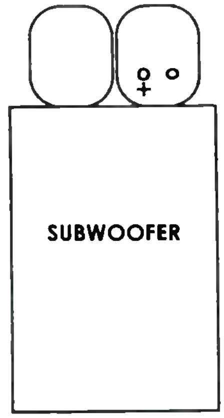

FIGURE 1: A typical dipolar or bipolar operating configuration.

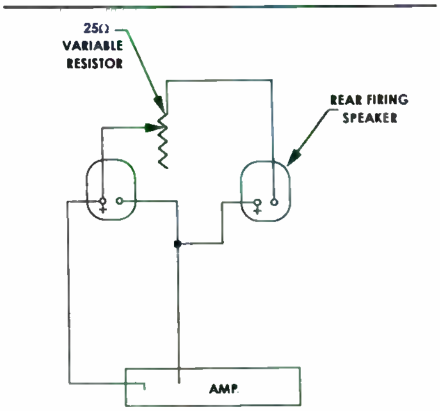

FIGURE 2: The author's variable resistor solution to control the output of rear-firing speakers.

My system features four Cambridge SoundWorks ambience speakers, with two facing forward and the other two paired with the first and facing backwards. The 25Ω resistor allows adjustment of the sound level of the rear-firing pair to compensate for speaker position and room characteristics. The four speakers reside two each on top of a pair of VMPS Smaller Subwoofers®, which fill in the lower octaves.

The result is a broad soundstage extending well beyond the outer edges of the speakers, a precise localization of performers across the soundstage, and a re-creation of a sense of the ambience in which the recording was made. This setup reveals deficiencies in a recording as well as its good attributes. It is also very revealing of microphone techniques.

The rear-firing speakers can be different than the front-firing ones—even in spectral balance, to a degree—and still give good results. The resistor enables you to make very fine adjustments to compensate for speaker differences. If you don't care for the effect, the system can easily be restored to its original configuration.

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SB *Mailbox*

POWER OUTPUT MIX-UP

Thank you for publishing my reply to Edwin Myers' query in *SB* Mailbox (*SB* 6/94, p. 57). I noticed an error in transcription, resulting in a confusing statement. The final sentence of the third paragraph should have read: for example, a 200W woofer with 100dB efficiency would match well with a 50W tweeter with 106dB efficiency, assuming proper level matching of components.

Bill Fitzmaurice
Laconia, NH 03246

TROUBLE WITH EQUATIONS

Thanks to Ed Dell and staff for publishing my article on the loudspeaker impedance curve ("Exploring Loudspeaker Impedance," *SB* 5/94, p. 28), which was motivated by the uneven showing of designs submitted for the A&S Speakers Sound-Off. The article's goal was to provide a bridge between material appearing in the *JAES* and the needs of the practical loudspeaker system designer.

Between my reading the publisher's proofs and the article's appearance in print, the usual gremlins surfaced to make random alterations to the copy. First, the expression EXP, appearing in the subscript of many equations, should not be subscripted. This is the exponential function, written EXP(x). I hope this causes no difficulties.

On p. 32, the equation following Equation (2) shows the independent variable X_P written in the numerator of a fraction. The X_P and the equals sign should be moved to the left of the fraction bar. Also in this equation, the slanted lines following three of the letters M are really fraction bars, as in Equation (A1).

Victor Staggs
Orange, CA 92667

MISPLACED EQUALIZER

In the 5/94 issue of *Speaker Builder*, D.E. Stenton states on page 18 of his "System III Loudspeaker" article that "the high-pass equalizer stage to each filter provides another phase inversion...."

Not so. As his Fig. 5 circuit on page 12 shows, the high-pass equalizer section is *noninverting*, and there is no reason to have it in the signal path feeding the high-pass filter.

The proper location for this high-pass equalizer is *after* the low-pass filter. The junction of R4 and R9 could then be used as the signal input.

Both the high- and low-pass filters could be configured as noninverting, unity-gain, third-order filters. Walt Jung's *Audio IC Op-Amp Applications* book shows how to do it.

In the 3/94 issue of *Glass Audio*, Fred Forssell mentions in his letter on page 43 that the serial data stream must be inverted at the input to the DAC if an inverting audio stage is used at the DAC output.

Burr-Brown's data on the PCM-58 and PCM-63 DA chips shows an inverting I-V op amp stage on the connection diagrams. This usually feeds a noninverting buffer output.

Utilizing the Curcio pre-preamp or a mu follower, both inverting circuits, along with a resistor for I-V conversion, should maintain proper absolute phase.

Tom Tutay
Ft. Walton Beach, FL 32549

D.E. Stenton responds:

Thanks to reader Tom Tutay for pointing out the text error describing the active equalizer and filter circuits. The high-pass equalizer is indeed a noninverting amplifier stage.

Although I intended to place the high-pass equalizer in front of the high-pass/low-pass filter stages as a buffer amplifier, in practice it should not be necessary to insert a buffer since the input resistance to the filter stage is twice the value of R9, or approximately 24kΩ. As Mr. Tutay correctly points out, the high-pass equalizer stage could follow the low-pass filter and the junction of R4 and R9 could then be used as the signal input.

To maintain proper phase at the input and output, the equalizer stage could be replaced by an inverting amplifier. The Burr-Brown op amp book, Operational Amplifiers—Design and Application, *shows an example of an op amp inverter circuit design. Readers who wish to develop a better understanding of these types of active filters should consult the article "An Electrically Tunable Low-Pass Filter"* (The Audio Amateur, *2/73). The discussion presented in this article can equally be applied to high-pass filters.*

Alternatively, you can replace the high-pass/low-pass active filters with noninverting unity-gain amplifiers, as indicated in Mr. Tutay's note.

Readers should also be aware of the incorrect unit that was specified in the last line of the last paragraph of the section describing calculations in Part 1 (p. 10). The value of the resistor in Fig. 7 is 0.5Ω, not 0.5W.

MORE POWER

In *SB* 6/94 ("*SB* Mailbox," p. 57), Edwin H. Myers, Jr. asked, "how much power will go to each component in a three-way system?" I would like to draw Mr. Myers' attention to an interesting and useful technical paper titled "Signal Power Spectrum Aspects in Loudspeaker Design" (*JAES*, Vol. 32, No. 9, September 1984, pp. 673–676).

This article, written by H. Mayr, provides

Come Out of the Closet

Fine craftsmanship seems to be in short supply these days, but fortunately, *Speaker Builder* has a large store from which to draw. Most *SB* readers are, by nature, craftsmen who have the ability to build beautiful equipment for their home-listening pleasure.

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a technical description to calculate the power division between drivers in a multiway loudspeaker system, as well as techniques for estimating the driver vibration amplitudes in actual use. As you would expect, the power split depends on the chosen crossover frequencies, and you can take this into account fairly easily using the data Mayr provides.

To cite an example presented in the paper, a three-channel system with the following channel limits, 20-500Hz, 500-3,150Hz, and 3,150-20kHz, will have a power split 72% : 27% : 1%. Mayr assumes that the signal is a wideband one with a continuous spectrum as defined in the German standard DIN 45 473, which was designed for checking the power handling capacity of loudspeakers.

Although the findings depend on the spectral characteristics of the assumed signal, they represent a good starting point for any analysis of loudspeaker power handling. I hope that Mr. Myers and others find these results to be of interest in their loudspeaker building endeavors.

Witold Waldman
Audiosoft
Melbourne, Australia

HISTORICAL PERSPECTIVES

"Loudspeakers: A Short History" by Gary Galo (*SB* 6/94, p. 18) is probably the most interesting article I have ever read in your magazine. Having worked with loudspeakers for the past 43 years, I was pleased to read Gary's effort to bring the younger set up to speed regarding the players in the history of quality loudspeaker development; otherwise, they might believe the plethora of acoustic junk that is on the market is all there ever was.

We should be most thankful to people such as Ed Villchur and Henry Kloss for developing (without computers, thank god) acoustic suspension technology. Ed May of JBL must be credited for his pioneering efforts with tuned port systems, even though I personally detest them. I was shocked and dismayed to see Henry Kloss purveying Band Pass subwoofers (?) in his Cambridge line. As Paul Klipsch would say, "sheer heresy."

Thiele and Small certainly contributed, but I submit their contributions benefited the software writers and sellers more than the average speaker builder. I carefully scan each *SB* issue hoping to read something above the technical masturbation on meaningless subjects. How many more times must we suffer through the intricacies of biscuit joints?

Speaker Builder should be about building reasonably simple, high-quality, and nonlunatic-fringe loudspeakers. In other words, put

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most of your contributing editors out to pasture and find some new blood that your mainstream subscriber can more easily relate to. In my mail-order business selling drivers and speaker builder parts, I regularly speak with *SB* subscribers, who are mostly looking for down-to-earth info they can comprehend.

Carl Roberts
The Speaker Works, Inc.
Phoenix, AZ 85014

Contributing Editor Gary Galo responds:

While I appreciate Mr. Roberts' compliments on my article, his comments on SB *articles in general are subject to debate. During its 15-year history,* Speaker Builder *has offered articles at the forefront of current loudspeaker mathematics and technology, and gained considerable respect in professional circles. Several of* SB*'s contributing editors and authors have presented papers on these subjects at Audio Engineering Society conventions, as well.*

One of this magazine's strengths is the variety of viewpoints and technical levels their authors address. It is important that SB *continue to offer a forum for differing opinions (such as yours and mine), as well as address the needs of readers of varying abilities—from beginner to advanced. I suggest that technically advanced articles, by well-respected authors in the field, lend greater credibility to the articles aimed at beginners, since they show a commitment to excellence at all levels.*

The ground-breaking work of Olson, Villchur, Kloss, and others is an extremely important part of our heritage. However, the work of their successors, including Thiele and Small, is equally important. Prior to the research of Thiele, Small, and Benson, loudspeaker design involved as much art as science. Today, vented and closed-box loudspeaker design is mathematically predictable to a degree not previously possible. The average speaker builder has benefited enormously from their work. Through the personal com-

PREVIEW
Glass Audio

Issue 4, 1994

- Inside the Russian Tube Industry
- Headphone Helper
- Balance in Push-Pull Amps
- Rectifying Your Drivers
- Purist Tube-Regulated Supply

puter and accompanying software, the nonmathematical builder can put such research into practice without an engineering degree.

Horn Study

Continued from page 36

size that close coupling to other drivers would be almost impossible. The AX2 has a cutoff frequency of around 750Hz, but is so smooth in its response that it can be used through cutoff. This is the lowest cutoff frequency that can be achieved, consistent with a flare which smoothly blends into the baffle, originating from a 1" throat in a diaphragm-to-mouth distance (with TD2001) not exceeding 300mm.

HORNS Q & A

However, a high-efficiency horn system, usable from below 1kHz to over 20kHz, with a 12½" mouth diameter, capable of producing very high fidelity and a maximum output of 125dB at 1m, is certainly a useful tool. And, it most definitely is not horn-like in its sound. Clearly, when the many variables are fully understood and appreciated, horn systems which do not possess typical horn-like vices *can* be produced. Attention to detail is a prerequisite, as is a comprehensive knowledge of the caveats.

Two further questions about horn design arise as a result of this research, both requiring further investigation:

1. Given the extreme sensitivity to small disturbances in the throat region, can the Tannoy concept of having an actual gap in the horn (the voice-coil gap of the bass cone) ever be expected to produce optimal results? More particularly, when that gap is modulated by high levels of bass driver movement, can a variable length, variable flare, gapped throat ever be expected to produce optimal results?

2. The results show that any abrupt flare rate changes within the horn can, and will, cause reflections that superimpose themselves on the transfer function. Since the concept of constant-directivity horns relies upon flare-rate changes of no subtle nature, can the best results ever be achieved from constant-directivity horns?[10]

Figure 5 shows the measured throat impedance plot of the AX2 compared to that of a widely used constant-directivity horn of reputable manufacture and similar dimensions. However, this should not overly concern manufacturers of constant-directivity horns, since the bulk of their sales are in the public address/sound reinforcement fields, where their smooth coverage of a desired area far outweighs the sonic subtleties discussed here. For studio purposes, though, constant-directivity horns would *not* seem to be the ultimate solution.

Grille Frames

Continued from page 48

frame, you are ready to use your jig to cut the sides. Scrupulously measure the outside dimensions that the frame will fill. Then cut the pieces so they are at least 2" longer than the finished dimensions. Clamp one piece with wedges in the right miter so it extends a bit past the center cut on the jig (*Photo 5*). The saw height should allow the gullet between the teeth to be at the surface. Slowly push the jig through, cutting the miter (*Photo 6*). If you encounter any difficulty getting the wedges out, you can use another as a driver (*Photo 7*).

Without turning the piece over, rotate the side to the left guide and align it so that the cut will produce the required finished length (*Photo 8*). Square frames require two sets of two pieces with the same dimensions. After you have made one piece and cut one end of the second piece, you can use the original piece to align the second piece (*Photo 9*). Repeat until you have cut all sides of the frame.

ASSEMBLY

I usually mark each joint (on the back side, of course), starting with one mark for the first joint, two marks for the second joint, and so on (*Photo 10*). If I didn't do this, I would mix up the sides and destroy a grain pattern that I have thoughtfully chosen (e.g., book-matched sets for opposing sides). Using a soft lead pencil, you can erase the marks after the frame is assembled.

The final assembly step is clamping the frame together and gluing and fitting it to the enclosure. We've already discussed gluing requirements and the woodworking tip that the shorter the time between cutting and gluing, the better the joint (*SB* 2/94, p. 46). Clamping is another subject we have discussed (*SB* 1/94, p. 58). The secret is applying the correct amount of pressure. If your frame is square, the jig described in *SB* 1/94 is one of the best ways to solve this problem (*Photo 11*).

You are now half done. In the next issue, I will discuss ways to dress up the frame, strengthen the joints, and attach the grille to your enclosure.

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McIntosh MC2002 power amp, mint, $1,050; Carver S7T mono amps, mint, $1,050; Soundcraftsman PP4 power amp, mint, $450; H-K Citation 11 preamp, excellent, $100; H-K Citation 12 preamp, excellent, $100; Philips 12" woofer mid tweeters, $5 each. Paul, (914) 528-1510.

Early Yamaha Audiophile level turntable: YP-800 direct drive turntable with YA-9 tonearm and Ortofon M20FL cartridge, includes disc washer and Shure stylus gauge, excellent condition, $100 or best offer plus shipping. Peter S., (207) 872-8577.

McIntosh stereo power amps: 2200, $1,000, two 2120, $650 each; MR-G5 tuner with multiplier, $450; Marantz Model 7 preamp, $1,350, Model 8 amp, $1,000; Counterpoint SA-20 amp, $1,000; Crown Macro Reference amp, Esoteric version, $2,250; JBL 2395 plate lens horns, 1" & 2" throat drivers. David, (914) 688-5024.

Pair Vandersteen 2As, original owner, need foam & grille, electrically excellent, $185; Fosgate 3602 surround processor with box & manual, mint, (originally $1,100) $225; D'Appolito originals 20ARS matched drivers, ¾" & 1" acrylic enclosures, separate back for crossovers, gray/black with matching S&D, don't like sound, $650 or trade. David, (310) 924-2666.

Altec-Lansing Model 15 speaker system, very good condition, 12" woofer, horn mid/tweeter with Tangerine phase plug, walnut veneer, $350 or best offer. Prefer local delivery or you pick up. Steve, (610) 759-3218 (eastern PA), leave message.

JBL: four 117H-1 8" woofers, factory sealed, $45 each; pair 2404H bi-radial horns, $250. Dynaudio: pair 21W54, $125; pair D-52AF, $95; pair 30W54, $175. Scan-Speak: pair D-2008, $40; pair 25W8561, $140; SEAS: pair H297, $15; four P14RCY, $20 each. Pair Electro-Voice ST350B, $115; pair Vifa D25AG-05, $25. Clay Jordan, (404) 969-8319, 6-10 p.m. EST.

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Total distribution	10,691	9,827
Office use, leftover	3,516	4,173
Return from news agents	126	0
Total	14,333	14,000

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WANTED

Single or pair Focal 7K011-DBL; Motorola T03 2N5686-5684. David, (310) 924-2666.

10V516 woofer; two AC-10 rubber surround drivers; three Variovents or equivalents; two 100W L-pads; two Scan-Speak D-2905 tweeters; two 12Mh Sledgehammer iron core inductors. Jim, (708) 425-6719, 5-10 p.m. CST.

Information on replacement voice coils for JVC DynaFlat ribbon tweeters and Radian/Emilar replacement voice coils. Want to buy Audio Precision test equipment. Dan Patten, 1768 N. 980 W., Orem, UT 84057, work (801) 224-8080 or home (801) 225-8577.

Service and/or owners manuals for Phase Linear model 700 power amp (direct-coupled solid state). Scott Roberts, 4533 N. Channel Ave., Portland, OR 97217, (503) 285-2904.

Knowledgeable speaker builder/audio entrepreneur with professional background in Quality Assurance and electronics seeks position or association with audio company in New York or southeastern states. P.M. Audio, (516) 868-8863.

Ad Index

Loudspeakers 101

MINDING THOSE Qs

By Dick Pierce

Last time, we explored a little of the history and philosophy behind the Thiele/Small parameters and looked at two of these parameters: resonant frequency and the "equivalent volume of compliance." Now we'll consider a class of parameters that, to many, are very mysterious in their nature but are just as important as the others.

SYMBOL OF QUALITY

To understand the Q parameters, Q_{MS}, Q_{ES}, Q_{TS}, and others, which don't seem to relate to everyday or intuitive experience, we need to take another little trip back into history. In the early days of radio, people were looking for a way to describe the quality of inductors, which were used, along with capacitors, to form the electrical resonant circuits that radios used for tuning and selectivity.

An inductor was considered high quality or "high-Q" if it had very little electrical resistance or losses compared to its inductance. It was considered "low-quality" if its electrical losses were high. Thus, the Q factor, or simply the Q, of a coil was a measure not so much of the quality in general, but specifically of the amount of energy that was lost to electrical resistance and, thus, could not be utilized for the resonant action of the circuit.

The higher the Q of a coil, the less energy was dissipated by the resistance of the coil. The result was a resonant circuit with a high-Q coil that exhibited a very sharp, narrow, and precise resonance, which is essential for accurate and selective tuning of radios.

ENERGY LOST

The Thiele/Small Q parameters are quite similar. They indicate how much energy is dissipated through different loss mechanisms in the driver. Two important losses or "resistances" occur in a loudspeaker driver: electrical and mechanical. Each of these Q numbers is a direct measure of the ratio of each of these losses to the total driver reactance at the driver's resonant frequency.

Unlike inductors used in radio frequency work, a high Q does not necessarily mean *better*. In fact, in some cases lower Qs indicate higher quality. In the case of loudspeaker drivers, a lower Q indicates better damping and control of the loudspeaker at low frequencies. Even so, an optimum range of Qs exists, depending upon application. Above or below this range may result in less than the best performance.

The mechanical Q, Q_{MS}, is a measure of the mechanical losses or "resistance" of the driver, primarily due to frictional losses in the suspension itself. It's very similar to the action of a shock absorber in an automobile suspension.

A lower Q_{MS} (in the range of 1–5 or so) generally indicates a lossy suspension, one that is good at terminating and controlling the flexing of the cone. However, this is only a very general rule, and there are examples of very high-quality drivers with comparatively high Q_{MS} values.

ELECTRICAL LOSSES

The electrical Q, Q_{ES}, is a measure of the electrical losses of the driver, primarily due to the simple electrical resistance of the voice coil. Usually, the driver's electrical Q is significantly lower than its mechanical Q, indicating that more energy is dissipated electrically than mechanically. That means the electrical Q is more important than the mechanical Q in controlling the speaker around resonance.

In practical loudspeakers, most of the electrical resistance in the loudspeaker circuit is in the voice coil. The total contribution of resistance from the crossover, speaker leads, and amplifier is almost always far lower and relatively insignificant compared to the voice coil. This contradicts the popularly held notion that reducing, for example, the speaker wire resistance from a tenth to a hundredth of an ohm can make a profound difference in how the speaker behaves at low frequencies (the effect is sometimes referred to as "damping factor"). But that's a story for another time.

In general, lower values for Q_{ES} indicate higher quality. You can infer that a lower Q_{ES} can mean a relatively larger magnet, for example.

MORE Qs

The total Q, Q_{TS}, combines mechanical and electrical resistance, and is a measure of the total losses that damp the motion of the cone. More importantly, it gives an idea about the suitability of a driver for a given application. As a general (but not inviolable) rule, Q_{TS} figures below about 0.40 indicate the driver is suitable for vented box applications, whereas closed box systems are well suited by drivers with a Q_{TS} ranging from about 0.35 to about 0.60. But these are only guidelines. There are certainly instances where a given application demands other values.

Another set of Q numbers relates to losses in the enclosure or vent. Q_A, for example, refers to the losses caused by any absorptive filling in the enclosure. Q_L refers to "leakage" losses in the enclosure. And Q_P is a measure of the losses in the vent or port due to air friction and turbulence effects. The combined measure of these losses is called Q_B. All of these enclosure Q values are usually a result of these losses measured around the system resonance (in the case of sealed boxes) or the enclosure resonance (for vented boxes).

We can also consider the "system" Qs, in the case of sealed boxes. For example, the Q_{TC} of a sealed box system describes the total system Q at resonance. In this case, the number directly relates to performance (since it describes the system as a whole). We can derive the response of a system from its Q_{TC}. For example, a Q_{TC} of 0.707 results in a "Butterworth" response, which is the flattest response possible. Q_{TC} values less than this indicate drooping, overdamped response, whereas greater Q_{TC} values indicate a peaked response and boomy sound.

A QUESTION OF DIMENSIONS

Let's take a little detour here. The Q parameters of a loudspeaker are referred to as "dimensionless," since they have no units. If the Q_{MS} of a loudspeaker was, say, 3, you might ask, "3 what?" Well, it's just 3, describing a ratio between two similar units (in this case, impedances, or losses).

Other units are "dimensioned" numbers, describing how much of something you have. For example, the V_{AS} of a speaker is always specified in liters, cubic meters, or cubic feet. The resonant frequency, f_S, is measured in hertz (cycles per second), or even radians per second. But the Qs, as with other dimensionless numbers, are nothing but factors or ratios.

I'll discuss some other examples of both dimensioned and dimensionless numbers in upcoming issues.

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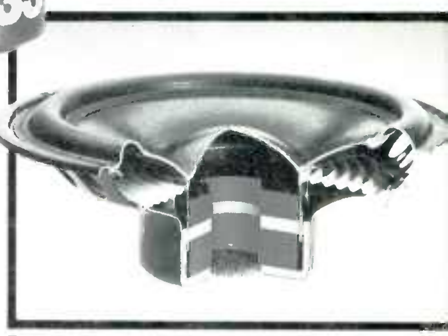

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