The Audio Critic®

Issue No. 19

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The pros love it. The tweaks tried to kill it. (See the loudspeaker reviews.)

In this issue:

We review nine loudspeaker systems from $829 to $8400 the pair, mostly very good and all very different, plus an amazing low-priced subwoofer.

The promised guest article on very high-efficiency speaker systems makes its belated appearance.

David Rich does his professorial number on still more preamps and several new D/A processors.

The David Ranada interviews with major thinkers in audio bring new insights in Part II of the series.

Plus delectable exposés of audio bull, tons of short CD reviews, letters to the Editor, and an entirely new column.

The Audio Critic

Issue No. 19 Spring 1993

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For subscription information and rates, see **inside back cover.**

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From the Editor/Publisher:
Whatever Happened to Fall 1992 and Winter 1992-93?

No, your mail didn't go astray. No, we haven't skipped two issues. This is Issue No. 19. The last one you received was No. 18. Only the dates are a little strange. Here's what happened:

Issue No. 18 was dated Spring/Summer 1992; however, it was delivered to our subscribers at the end of summer, in mid-September, very close to the beginning of fall. It would have been unrealistic to date the next issue Fall 1992 and the one after that Winter 1992-93 because even in the ideal case (meaning: without our usual delays) we would have been stuck in an almost-next-season pattern forever. We were therefore going to call No. 19 the Fall/Winter 1992-93 issue and try to publish it at the beginning of winter, still not much more than a 3-month interval after No. 18, but that didn't quite work out, either.

Further delays were caused by a number of desirable but time-consuming new developments. We had to organize our distribution at newsdealers, bookstores, audio stores, and other retail outlets, this being our first issue with coast-to-coast retail distribution. On the advice of our distributors we made some changes in the appearance of our cover pages for greater recognizability and sales appeal on magazine shelves with overlapping rows of publications. In consequence of our larger print run, we had to make small changes in our page dimensions and other specifications in order to permit printing on a web press. Finally, we had to get ready for second-class mailing for the first time in our history; on top of it, our authorization to mail at second-class postage rates was delayed way past all deadlines.

Doing things differently for the first time always takes longer than expected; add to that our old—should I say traditional?—lateness problems due to the lack of a full-time staff, and here we are again—late winter as I write these lines. So, for the reasons already given, Fall/Winter 1992-93 is out and Spring 1993 is the only realistic designation. That at least shifts us to the beginning of the season and gives us a fighting chance to have Summer 1993, Fall 1993, etc., issues whose publication actually corresponds to those dates. As I told you last time in this same space, I have given up making promises; all I'm willing to say is that I don't expect the nonrecurrent problems discussed above to cause new delays.

Box 978

Letters to the Editor

We get quite a few intelligent, well-informed, and well-written letters that should never have been sent and will never be published in this column. Why not? Because they ask questions and bring up arguments that have already been answered in our pages, usually in the very article or review that elicited the letter. It's amazing how many people would rather write than read. Halfway through the article they get excited and rush to their typewriter or word processor. Please read what we have to say, every word of it, before attempting reciprocal punditry. Letters printed here may or may not be excerpted at the discretion of the Editor. Ellipsis (...) indicates omission. Address all editorial correspondence to the Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951.

The Audio Critic:

We were amused by the letter on page 8 of Issue No. 18, in which the ridiculous suggestion was made that neither magnetic fields nor vibration have any effect on the performance of an audio system. *[That is indeed ridiculous but very far from what the letter actually said. See my reply below.—Ed.]* This type of misunderstanding is to be expected from a nontechnical reader, but we found it surprising that the Editor agreed with him. The letter must have been a last-minute insertion. *[A sarcastic reference to my lame excuse for letting a tweako ad slip through.—Ed.]*

We at MSB Technology have designed *all* of our products on the basis of solid engineering principles, including our EMA Isolation Plate. At the risk of boring the more technically astute readers, we will review the basics of motion, current, and magnetic flux. This review clearly shows why magnetic fields and vibration affect sonics, and why such a plethora of products has been developed, either by chance or design, in this field.

We will start with the simple equation

$$\boldsymbol{\Phi} = \boldsymbol{F}/\boldsymbol{R}$$

where $\boldsymbol{\Phi}$ is magnetic flux, $\boldsymbol{F}$ is magnetomotive force (remember also that magnetomotive force $= 0.4\pi NI$, where $N =$ number of turns and $I =$ current), and $\boldsymbol{R}$ is reluctance (magnetic resistance).

One cannot help but see that changes in magnetic flux will change magnetomotive force, and on the other hand changes in current will change magnetomotive force and thus magnetic flux. The principle is the basis for power generation, motors, and of course all speakers. In ribbon speakers, for example, as the current changes in a film conductor within a magnetic field, the conductor moves, creating sound. So current in a conductor in a magnetic field can cause the conductor to move.

The same principle can also be reversed. When a conductor is moved in a magnetic field, a current is induced in the conductor. This is how generators work, as well as early microphones and record players. As the needle vibrates in the record groove, a magnet is vibrated, causing the magnetic field to move relative to the conductor, generating current in the conductor—the audio signal. In any audio product, as a conductor carrying an audio signal is externally moved in a magnetic field, extraneous currents are induced within that conductor! These currents add to or subtract from the audio signal.

At each end of your most basic audio system are clear examples of the role of vibration and magnetic fields in playing back and generating sound. Within an audio component, such as a CD player, are many interesting sources of magnetic fields, including transformers, drive motors, and servos. Many of theses sources change dramatically during playback, while others are more predictable. Amplifiers have very large transformers and fields. Fields are even created around individual wires and components. If anyone is interested in a detailed characterization of these sources, an excellent source book for further study is Volume 8 of the Handbook Series on Electromagnetic Interference and Compatibility. Many sources of vibration also exist, including direct feedback coupled through the air or floor from the speakers, drive motor and servo vibration, and magnetically induced power supply vibration (ever felt a large power transformer hum?).

At MSB Technology Corporation, we understood these basic principles and have created the EMA Isolation Plate to best deal with both the vibrational problems and the magnetic field problems.

Let's start with magnetic fields and a simple illustration. Magnetic field lines radiate out into space from a horseshoe magnet. (Remember the iron-filings-on-a-sheet-of-paper experiment in grade school?) *[I know when I'm being patronized, and so do our readers.—Ed.]* When an iron bar is brought into proximity of the magnet, the field finds less resistance in the iron, and the field is coupled through the iron. It no longer radiates into space. In the same way an EMA Isolation Plate, on one side of a large magnetic field, can couple that field. A simple experiment with a magnet will illustrate this principle.

As you can see, a solid iron plate would be totally effective in coupling magnetic fields. This is a problem, however. The magnetic sources are electrical, and losses will occur in a solid plate. Excessive losses could mean reduced performance of the component. (We observed this with CD player design). Eddy current losses and hysteresis losses are the principal source of loss in magnetic coupling. Hysteresis losses are primarily material-dependent, with the Steinmetz coefficient providing an indication of the quality of the material. (An 8 to 1 difference in performance is seen just among different types of steel). Eddy current losses are given by

$$P_e = (\pi t f B)^2/6p10^{16}$$

where t = thickness, f = frequency, B = flux density, p = resistivity. Notice that by reducing sheet thickness, losses are reduced. This is why transformer cores are made of laminated sheets—to reduce eddy current losses. This is also why the EMA Isolation Plate is made up of thin layers of low-Steinmetz-coefficient material. The plate is designed to couple magnetic fields with a minimum of loss, or external load on the component.

The second principle is vibration control. Simplifying a discussion of vibration is difficult. A more rigorous discussion on the subject can be found in the *Theory of Vibration with Applications* by William T. Thomson. As vibration in an object is damped, the amplitude of each cycle of vibration is reduced. The ratio of the amplitude of each cycle to the next cycle is called the amplitude ratio and is

$$x_1/x_2 = e^{\delta}$$

where δ is the logarithmic decrement

$$\delta = 2\pi\zeta/\underline{\sqrt{1-\zeta^2}}$$

where ζ is the damping factor

$$\zeta = c/(2mw_n)$$

where c = a constant of proportionality, m = mass, and w_n = the natural frequency of the system.

At the risk of becoming too complex, it is important to understand that primarily two elements determine the vibrational damping of a system. First is the system mass. Mass determines the natural frequency w_n. Second is the damping factor ζ of the isolation material. With these simple tools in hand one can evaluate the most outrageous "vibration control" product, from clamps to gooey stuff, and evaluate its contribution to mass and damping factor. The EMA Isolation Plate weighs about 50 lb. and is damped with Isodamp, a superb material produced by E-A-R Corporation, with a damping factor of 0.6.

Finally, we have demonstrated the theoretical importance of vibration and magnetic fields, and have shown how the EMA Isolation Plate directly controls both. The last test is subjective evaluation. We have found the EMA Isolation Plate is effective in improving sound quality under a wide variety of products. Its effectiveness depends on the degree of magnetic shielding in the component, the component's mass and isolation material, and its proximity to other components. Most electrical engineers have little understanding of shielding and vibration, unless experienced in RF and microwave applications. Most audio products are inadequately shielded and damped.

At MSB Technology Corporation we have utilized the EMA Isolation Plate in all our products. We have used it as a base to build our CD player and transport on for many years. Recently, we developed the MSB Processor, which is built entirely within an EMA Isolation Plate. We hollowed out three sections within the core of the plate, one for the power transformers, one for digital, and one for analog. Our circuits are potted within the core for complete isolation and vibration control. This unique product has outperformed every other D/A it has been compared with.

We hope this simple tutorial will help you and the reader better understand the important role vibration and magnetism play in high-end audio. I can be reached at (415) 747-0271 if anyone wishes to discuss this topic in more detail.

Larry S. Gullman,
B.S.M.E., M.S., P.E.
General Manager
MSB Technology Corporation

It seems fairly obvious to me that you're just playing games here. Your opening game is to restate and alter what the letter writer (Graham Ross of San Mateo, CA) was actually saying, in order to make it sound absurd. Read his letter again. He never said that magnetic fields or vibration will have no effect on an audio system, period. He said, in essence, that your particular product is unlikely to have an effect. Quite a difference.

After that straw-man game, you play the we-engineers/you-laymen snob game. Well, we don't know whether or not Mr. Ross is an engineer—he could be one, couldn't he ?—but I have at least as many engineers in my corner as you do, so I'm not impressed.

Then you proceed to your big steamroller game. You cut-and-paste standard formulas from a reference book and offer that as scientific proof of the effectiveness of your EMA Isolation Plate. It's a classic non sequitur, about as logical and convincing as saying, "$E = mc^2$, therefore cold fusion works." (Come to think of it, those Utah cold-fusion fantasists at least reported some measurements.) We all agree about the laws of electromagnetism, vibration, etc. Citing them doesn't prove a specific product claim. Instead of trying to intimidate your opposition with Greek letters, you should have shown what kind of signal comes out of the back end of a CD player, first without and then with your Isolation Plate. If you can show a significant difference in a typical audiophile environment when all other conditions are equal, then I'll begin to believe. So far you haven't proved anything. I still think the EMA Isolation Plate is a high-end marketing gimmick, not a solution to a real-world problem. The fact that there's a highly qualified technical team behind it doesn't change my perception.

As for your "simple tutorial"—it simply isn't one. A simple tutorial would (1) define all terms and (2) leave the constants out of the basic formulas. You were deliberately trying to be complicated rather than simple, to be difficult rather than easy to understand, for purposes of professional intimidation. I may not be a professional engineer, but I can tell a stratagem of adversarial dialectic when I see one.

—Ed.

The Audio Critic:

Many thanks to you and Dr. David Rich for including the Sumo Athena II in your exhaustive preamplifier survey in Issue No. 18. In reading the review, it was clear *The Audio Critic* has a unique understanding of the technical details of preamplifier design, so we were very excited the Athena II was one of models recommended to your readers. We would like to make the following comments:

(1) The Athena II chassis, subpanel, and top cover are made of 20-gauge steel. The knobs, rack mounts, and front panel are solid aluminum. The front panel is not made of plastic.

(2) Some running production changes were made to the Athena II [around February 1992]. The PC board is now double-sided with plated-through holes. Also, capacitors in the primary signal path have been eliminated, and the line gain stage uses a DC servo. By the way, the front panel is not made of plastic.

(3) Sumo has never manufactured, considered, thought of, looked at, dreamed of, or even been in close proximity to a front panel made of plastic. The Athena II front panel is a high-grade custom aluminum extrusion that is machined, brushed, and black-anodized.

(4) There is a remarkable amount of substance to Dr. Rich's presentation of preamplifier design. Once we recover from the ignominy of having our pride and joy described as having a plastic front panel, we look forward to making further technical comment.

Okay, I guess we have to admit that's all the self-righteous indignation (our tongue held firmly in cheek) we can squeeze out, since we try not to take ourselves too seriously. After all, this business is supposed to be about having fun listening to music.

Sincerely,
Michael Custer
President
SUMO
Agoura Hills, CA

David Rich never, never listens to anything but classical music. Don't expect him to know about heavy metal—not even in front panels.

—Ed.

The Audio Critic:

To begin with, I'd like to express my enjoyment of *The Audio Critic* in general and of the latest (No. 18) issue especially....

I do have a problem with a couple of areas in Dr. Rich's otherwise excellent review of preamps. The first is his refusal to test tube preamps. The "buggy whip" analogy is flawed. *Car and Driver, Road & Track* and other "high-end" auto magazines have reviewed the Dodge Viper, the Corvette, the "new" Shelby Cobra, and similar autos which are based on technology as "new" as tubes are. I would argue that the high ratings given to these cars are for the same reasons that tube electronics are still highly rated in audio circles. To simply "write off affordable preamps from Conrad-Johnson, Van Alstine, Joe Curcio, and Sonic Frontiers tends to show more about the reviewer's prejudices than whether such preamps are competitive with those reviewed. This detracts from an otherwise fine article. Two, I think it is important that when a reviewer implies the use of "subjective" listening sessions in a review that the entire setup be listed. The cartridge/arm/turntable combination is especially critical when evaluating how well the phono section of a preamp works. The amplifiers and speakers are also a useful tool in evaluating how a particular component will sound with other components. Finally, does Dr. Rich have a problem with Adcom? He implies that because the Adcom GFP-565 uses linear switches it is flawed, or that this contributes to the overall dissatisfaction with the unit. Yet he implies that in the Sumo Athena II the same switches are fine. In fact, the same Alps rotational-to-linear control/switch assembly is used in both preamps.

Overall the review was very informative and enjoyable. Dr. Rich has a very clear writing style and his article was never a chore to read. I look forward to more....

Once again let me thank you for such an interesting publication. You are able to strike a balance against the "far left" of audiophiles without resorting to name-calling or paranoia. I look forward to the next issue.

Sincerely,
Michael Baker
Falls Church, VA

Much as I appreciate your compliments, I have a sneaking suspicion that you still have one foot (or at least a few residual toes) in that left-leaning camp of audiophiles, otherwise you wouldn't be saying some of the things you put in your letter.

Whether or not the buggy-whip analogy is apt is rather beside the point. (As an erstwhile car buff, I could also question your choice of analogous automobiles, but this is an audio magazine.) The point is this: the vacuum tube is an outmoded device, at least for audio applications. Any audio circuit that can be done with tubes can be done better, or at the very least just as well, with transistors. There is simply no credible technical reason to go the tube route. I'm willing to concede that a faultlessly engineered tube preamplifier is essentially as good and useful as a similarly well-engineered solid-state preamplifier, but the tube preamp will lose performance as the tubes age, and even when new its distortion plus noise will never be as low as 0.002%. Why do it, then? The fact is that vacuum-tube audio circuit design has little or no support in the professional engineering community; all, or nearly all, the tube amplifier companies are owned and run by tweaky audiophiles without engineering degrees, and the rave reviews come from their groupies at the tweaky magazines. What drives the tube amplifier market is a belief system, not a superior technology.

As for critical listening setups, mine changes (of necessity) all the time, and David's also changes fairly often. Of course, we would never use any components we have found fault with, but we don't get as dogmatic about reference systems as the tweako reviewers and their readers because, unlike them, we perform subjective listening tests for confirmation and verification, not for revelation. When the specifics of the listening setup become critical for some reason, we do get specific.

Lastly, your skeptical comment on the Adcom and Sumo reviews is based on totally false information. The switch assembly in the Sumo Athena II sample we reviewed was made by Nobel, not Alps as you incorrectly believe. As pointed out in the review, the Nobel rotational-to-linear converter is quite robust, whereas the Alps unit used in the Adcom GFP-565 is much less so. Perhaps more important is the fact that the rotational-to-linear converter is connected directly to the switch in the Sumo but indirectly through a long unsealed band of metal in the Adcom. Rest assured that ***The Audio Critic*** *and its staff don't "have a problem" with any particular brand—unlike some publications I could name, we genuinely don't care who ends up looking good in our tests and who doesn't.*

—Ed.

The Audio Critic:

Enclosed please find an *IAR Hotline!* article (61-62, Aug. 1991) touting the theoretical and "observed" superiority of multibit D/A converters over delta-sigma D/A converters. Since I do not have the technical sophistication to see what is wrong with Mr. Moncrieff's argument, I would be delighted if you or Dr. Rich would explain what is wrong in a future *TAC* article...or published letter. Please ...enlighten me with a short explanation or direct me to an explanation.

Thank you.

Peter Bandurian
Louisville, CO

The article you enclosed is too long to be reprinted here, even if there were no copyright problems, and too inconsequential to be refuted in a full-length article of our own. The question it raises, however—is delta-sigma a comedown from multibit?—keeps cropping up in our mailbox and deserves an answer.

As usual, Peter Moncrieff starts out with sufficient technical understanding to give him an entry-level grasp of a complex subject—for which he keeps patting himself on the back—and then boldly sails his ship into tweako waters because his understanding is incomplete, alas. I asked our Technical Consultant, Steven Norsworthy, who is a recognized IEEE authority on digital signal processing in general and delta-sigma converters in particular, to comment briefly on Mr. Moncrieff's errors and on the true nature of delta-sigma conversion. His somewhat technical exegesis follows; if you find it a bit abstruse, just cut straight to his conclusions. Delta-sigma DACs have their potential stumbling blocks, as has been pointed out in these pages before, but lack of 16-bit resolution definitely isn't one of them.

* * *

With regard to Mr. Moncrieff's article on **ΔΣ** (delta-sigma) converters, there are some extremely obvious flaws. In an attempt to make **ΔΣ** converter theory accessible to the layman, the popular audio press has presented simplistic models which do not even begin to describe what is actually going on inside such a converter. This is tantamount to misinformation. So, nonengineers like Mr. Moncrieff try to make sense out of this nonsense, which results in jumping to false conclusions. Unfortunately, it seems that Mr. Moncrieff has not consulted with professionally degreed and experienced engineers who are knowledgeable on the subject.

The **ΔΣ** D/A converter (DAC) process begins in a manner similar to that of an ordinary multibit DAC. The sampling rate of the 16-bit 44.1 kHz PCM-encoded signal is increased, typically by a factor of 8, up to 352.8 kHz. This causes the frequencies between 0 Hz and 22.05 kHz to repeat at multiples of 44.1 kHz all the way up to 176.4 kHz, which is one half of the new sampling frequency. These repeated frequencies are commonly known as spectral images, and they are removed by a digital lowpass filter. This whole process is known as interpolation. Typically, the data in the digital lowpass filter is processed with greater than 16-bit accuracy so that the net accuracy of the result is not less than 16 bits. Usually the output words are 18 bits, sometimes 20 bits.

Now, the next steps that follow are different for the multibit DAC versus the **ΔΣ** DAC. The multibit DAC takes these 18-bit words at 352.8 kHz and converts them directly to a corresponding analog voltage level. This means that the DAC must be capable of generating 2^{18}, that is 262,144 levels. Following this, an analog lowpass reconstruction filter removes spectral images, which repeat at multiples of 352.8 kHz.

In the case of the **ΔΣ** DAC, the 18-bit 352.8 kHz output of the digital lowpass filter has its sampling rate increased further, by as much as 32 times that rate, to 11.3 MHz, so that now the word rate is as high as 256 times the original 44.1 kHz rate. This signal is fed into what is known as a digital **ΔΣ** modulator. Within the modulator, the noise introduced by a 1-bit quantizer is digitally highpass filtered so as to suppress noise in the signal band from 0 Hz to 22.05 kHz, while the noise above 22.05 kHz is greatly increased. This suppression of inband noise is what enables the converter to be capable of achieving its ultimate resolution of 16 or more bits. An analog lowpass filter then converts the 1-bit output of the modulator and removes the high-frequency noise. In order to assure the preservation of the original 0 Hz to 22.05 kHz signal, one only needs to measure the signal-to-noise ratio in this frequency band from a power spectrum, which is relatively straightforward to do with proper laboratory equipment on an actual DAC system.

It is interesting to note that the actual information content is much greater for a 1-bit signal at 256 times the original sampling frequency than for the original 16-bit signal. In simple terms, there are 256 bits of information for every original 16-bit sample! (This is completely contrary to Mr. Moncrieff's claim. He says there are 256 possible *levels,* which are represented by only 8 bits, i.e., 2^8 combinations. There are actually 2^{256} possible combinations of 256 bits!) Of course, much of this information is the high-frequency noise introduced by the **ΔΣ** modulator and is somewhat unrelated to the original 16-bit input samples.

In summary, there are no technical reasons to doubt that a properly designed **ΔΣ** DAC system is any less capable of 16-bit (or greater) resolution than a comparable multibit system. This is *not* to say that all commercially available **ΔΣ** DACs are flawlessly designed. Indeed, some DACs that are advertised as having 16-bit quality have subtle flaws inherent in either the basic architecture, or the actual practical implementation, which can cause loss of resolution. I plan to address this subject in a forthcoming issue of ***The Audio Critic.***

—Steven R. Norsworthy

* * *

As for those funny waveform traces shown in the article, they illustrate something altogether different from what Mr. Moncrieff thinks; they are the necessary consequence of the different analog filter configurations used in the different pieces of equipment, with different rise times, different ringing characteristics, etc. They have nothing to do with the DACs.

I must add that I, personally, am not necessarily more "sophisticated" in such purely technical matters than Mr. Moncrieff, but it seems that I associate with more sophisticated practitioners than he does. I must further add that none of us here would ever dream of referring to our publication as TAC; *that kind of clubby alphabet soup is in the style of the lightweight tweako journals.*

—Ed.

The Audio Critic:

...I have often wondered how many impending improvements to equipment that manufacturers disclose to editors result from a wish to deflect criticism gracefully and how many were actually contemplated before the product's deficiencies were pointed out by journalists.

[On another subject], on page 13 of his most interesting and valuable article [Issue No. 18, "Reasonably Priced Preamplifiers..."], David Rich refers to an LP record that is "no longer in print." You are implored not to allow this inappropriate expression to creep into your fine journal. Since LPs and CDs are not printed in the first place, when no longer available are properly referred to as deleted, which of course describes their status in their respective catalogues. If this seems too arcane, we are willing to give conditional approval to "discontinued" or "unavailable."

I only point this out because the penalties for such faulty terminology can, in certain jurisdictions, be quite severe...

József Izsák
Toronto, Ont., Canada

/ often tell David Rich, who always wants to redesign the electronic equipment he is reviewing, that our job is to evaluate what exists, not to find better solutions for the manufacturers. It is true, however, that some manufacturers look to the audio journals as a source of design guidelines, sometimes regardless of the reviewer's credentials. Each case is different, so I can't make wholesale generalizations here, but a negative review by, say, Bob Harley shouldn't make a graduate E.E. change his carefully designed circuit just to get a better review next time. That would be sad.

*The lapse in David Rich's terminology is my editorial oversight; his copy comes to me replete with the (ahem) stylistic impurities (ahem) that an American engineering education tends to leave unremoved. Your admonition is well-taken and will be heeded. The funny part of it, Jóska, is that it seems to take at least two Hungarians to keep **The Audio Critic**'s English out of trouble.*

—Ed.

The Audio Critic:

In recent editorials in *Stereophile,* Robert Harley has suggested that technical performance is overemphasized when reviewing audio components. Instead, we should concentrate on the equipment's ability to convey the "emotional content" of the music. Imagine if we applied that logic to other types of reviews:

"I was moved to tears by the hardbound edition of *Gone with the Wind,* but the paperback left me cold."

Or:

"I laughed hysterically watching *The Simpsons* on the 27" Sony, but the 26" Toshiba rendered the same episode dull and lifeless."

The quality of a device designed to convey information must be assessed using parameters that measure the accuracy of the information conveyed. The emotional content of the subject has little relation to the design of the typeface or the quality of the picture—it's inherent in the subject itself! And as long as it's accurately presented (measured using parameters such as legibility, color contrast, etc.), the inherent emotional content remains unchanged. Just as television sets are measured on their ability to accurately present a picture, audio components must be evaluated based on their ability to accurately present the sound. Nothing more, nothing less. Robert Harley would do well to leave the emotion to the music.

Mark L. Swierczek
Great Mills, MD

/ not only agree with you but have editorialized in the same vein more than once. (See, for example, Issue No. 17, page 45, first indented paragraph.)

Of course, Robert Harley can't be objectively tested on what he feels *when he listens, the way he could be tested on what he actually* hears *(if he didn't refuse to be so tested). Thus, for all we know, he is moved to ecstasy by certain logos and front panels, and is left cold by others. That's entirely possible. We do know that he must look at the brand name on the front panel before he is able to form an opinion of the sound.*

—Ed.

The Audio Critic:

I have received every single issue of your publication, and I very much want you to continue publishing. You are the only magazine that I trust to give me the necessary facts to make an intelligent decision about what audio equipment to buy. I am totally unable to spend the kind of money other high-end publications recommend, and I have a natural distrust in someone who thinks spending $10,000 for a preamp will make them happy.

Many of my components have been purchased after reading about them in your magazine. For instance, I still enjoy listening to LPs through an Advent Model 300 receiver. It is not my main preamp, but it still works well.

I also receive almost every other audio publication around, as I am not only a musician but a music lover, and I like to keep up with the latest in equipment even though the newest component I purchased is over two years old. Yours is the only publication whose reviews discuss the practical reality of design features (and flaws). I really get tired of reading about how the cymbal crash at measure 149 in Mahler's Third sounded with this speaker, or that amplifier, or how the hushed murmur of the strings moved this reviewer or that one. I want to know what those unbuffered tape outputs mean if I buy that preamp, and you deliver when it counts. Keep up the good work.

Charles Hardgrave
Goldthwaite, TX

*I gratefully acknowledge and genuinely appreciate your kind words, but let's not go overboard here. A competent audio component review is necessarily focused on the engineering of the unit, but the sound of that cymbal crash (or of those murmuring strings) can be an important clue to an engineering fault or advantage, especially in the case of loudspeakers. Self-indulgent subjective reviewing is worthless, but technical analysis without listening is also of rather limited value. We at **The Audio Critic** measure everything, listen to everything, and try to get the best professional opinion on the engineering strengths and weaknesses of everything—because only such a comprehensive approach meets our standards of accountability.*

—Ed.

The Audio Critic:

Congratulations on the addition of David Ranada to your staff. His interviews were well focused, and I am looking forward to more of them. I particularly appreciate Kevin Voecks's statement that loudspeaker models of the same type should not vary in quality.

The letters column in Issue No. 18 discussed your modified views on the importance of absolute phase. Another point made in an early issue of *The Audio Critic* is that electronic components need to warm up for at least an hour before they sound their best. Do you still believe this? I once did but no longer do.

The only CD player that you have found to be audibly superior to its competition was a Sony unit modified by Precision Audio. I am curious why you have not used this unit (or another Precision Audio modified model) as reference in your more recent tests of CD players and D/A converters.

Keep up the great work!

Sincerely yours,
David Altman
Maywood, NJ

/ must confess that we haven't so far investigated the warm-up question scientifically. The right way to do it would be

to go through our complete measurement protocol on a dead-cold unit immediately after turn-on, then again after a one-hour warm-up, and once again after 24 hours. It could very well be that the differences are trivially small or nonexistent, but just in case I'm wrong I always warm up the equipment for at least 30 minutes before any serious listening. (I also knock on wood occasionally.) One of these days I'll do the tests, at least on the components that are semipermanent in my system. (No, I don't trust my earlier subjective perceptions on this anymore.)

As for the Sony CD player modified by Precision Audio (New York), they took it back from me shortly after my review. (Incidentally, I corrected their name in your letter; you wrote Audio Precision, which is a much larger company in Oregon, making state-of-the-art audio measurement gear.) Many of the more recent CD players and DACs reflect the same circuit design philosophy as the Precision Audio mod, so I don't think I lost an irreplaceable reference.

—Ed.

The Audio Critic:

...I have struggled over the years, desperately trying to hear the "astounding" differences attributed to various system tweaks by the high-end community. I always wanted to hear these differences so that I could justify spending large sums on beautiful toys. I have usually failed to hear these "differences," so I made purchasing decisions on the basis of friends' and trusted dealer recommendations.

I now find myself with a decent system... *[The letter lists the very respectable components in the system, but they are of no relevance to the issue raised below.—Ed.]*

I genuinely enjoy the sound of my system but have been anxious to have remote control, at least of volume, from the listening chair. This idea has seemed to be anathema to the high-end community. I cannot understand why. I was excited to hear about the Forte 44, but then heard from my local dealer that since the reorganization at Threshold he no longer plans to carry the line because he was disappointed in the way the company has put together the product.

In your most recent preamplifier survey, none of the units had remote control. I would love to hear some discussion in the magazine about why only Jeff Rowland and Krell seem to be able to offer "properly done" remote control, and also the opinion of you and your editors about the possible availability of such units in the near future.

Thanks.

Yours truly,
Eric Brody
Lake Oswego, OR

Remote control has never been important or even mildly interesting to me because I am a pacer rather than a couch potato. I jump up, pace around, go to the front panel, adjust the volume or select another source, pace around again, and sit down only later. The finer points of remote control are therefore beyond my ken, so I asked Dr. Rich (who is even more hyper than I am but at least knows the subject cold) to answer you.

* * *

Two distinct problems must be solved in a remote-control preamp. The first is the switching of signals. One method is to use a CMOS solid-state switch. CMOS switches restrict the maximum input swing of the switched signal to some fraction of the power-supply voltage of the switch. This is typically ±5 V, although some expensive switches allow more that that. If the input signal exceeds the power supply of the switch, the switch can be driven into a potentially destructive latchup mode. Another problem with a CMOS switch is that it has a relatively high on resistance. This, in conjunction with the nonlinear junction capacitance of the switch, can lead to signal distortion. The on resistance of the switch is itself nonlinear, and it is critical not to have the CMOS switch terminated into a resistive load. Bipolar devices are not usually used for switches because the voltage drop, $V_{CE(sat)}$, across a bipolar device cannot be brought to zero. Another method of performing switching is to use a relay. There can be reliability problems caused by the mechanism of the relay or by corrosion of the electrical contacts due to an insufficient wetting current. Sealed relays with switch contacts made of rare earth materials can minimize these problems, but these devices are expensive.

The second problem involves the design of the remote-controlled volume control. One method is to use a tapped resistor string, with the desired tap selected by a CMOS switch or relay. The aforementioned problems associated with these components will then arise. A second method is to use a voltage-controlled amplifier (VCA). The VCA is an electronic circuit in which a DC control voltage sets the voltage gain of the circuit. These circuits often have limitations in dynamic range and have relatively high distortion. Those interested in a detailed analysis of VCAs are referred to Ben Duncan's series in *Studio Sound* magazine. The third method is to drive a standard potentiometer with a servo-controlled motor. This solution extracts no performance penalties.

—David Rich

* * *

I'm pretty sure those cute little servo-controlled motors ordered in limited quantities aren't cost-effective for small audio manufacturers. It seems remarkable, therefore, that Forte puts the industry's first motorized input selector in the 44 at the $1095 price point, whereas Krell still proudly advertises relays for the switching functions in the KRC at $6000 and motorized control only for the volume. I believe that Jeff Rowland also uses relays for the switching functions. Apparently there's always more than one "properly done " engineering solution. (Incidentally, David Rich was unaware of Forté's motorized input switching when he wrote the above but later said it sounds like a good solution to him.) As for your local dealer, how do you know whether he dropped Forte or the new Threshold organization dropped him ?

—Ed.

Erratum

Our review of the Acurus L10 preamp, printed as an advertisement apparently contained an error. According to Mondial Designs, William Snyder is not "now a consultant to Mondial." He did, however, design the basic circuit of the Acurus L10, whereas the actual hardware implementation was the work of Mike Kusiak. Our rule is that advertising reprints must be exactly what was published, warts and all, hence the form of this correction.

Divergent Design Philosophies: Nine Speaker Systems and a Subwoofer.

By Peter Aczel
Editor and Publisher

While the electronic components in the audio chain are gradually converging toward a few basic design architectures, audiophile-quality loudspeakers are still trying to make highly individual statements. Is that good? Read the reviews and decide.

Loudspeakers have been my number one topic of interest since the earliest days of ***The Audio Critic;*** I have written on the subjects of speaker design and speaker testing at great length; longtime subscribers know exactly where I stand, and newcomers are finding out rapidly—so I have no intention to repeat myself here just because a new crop of speakers came in for review. Readers are referred to Issues No. 10, 11, 14, 16, and 17 for background information about my approach and predilections.

The only new thing I want to note here is that I have started to make use of the MLS (Maximum Length Sequence) testing capability of the Audio Precision "System One Dual Domain" in order to obtain the equivalent of anechoic response measurements above approximately 300 Hz. (The more widely used and much less costly MLSSA add-on system for the PC is based on the same principle.) I can't say that it has been a revelation; my older, cruder methods also showed me what I needed to know, albeit less conveniently and a little less accurately.

At any rate, since we don't really have a mathematical model for the "perfect" loudspeaker operating within room boundaries—as we do, e.g., for the "perfect" amplifier—all loudspeaker testing remains a bit tentative and open to argument on certain points, although we can readily (and incontrovertibly) measure most of the important performance characteristics. The measurements invariably show nontrivial differences between input and output—i.e., distortion—in all speakers, and that means (1) that we don't know what a totally nondistorting speaker would sound like and (2) that the preferred choice between one kind of distortion and another is necessarily subjective, at least to some degree. The soon forthcoming DSP-corrected loudspeaker systems will perhaps begin to change that situation, but don't count on amplifier-perfect speakers just yet.

It is interesting to note that, even though loudspeakers are obviously the one remaining component category open to insufferably self-indulgent subjective reviewing, most of the insufferable subjective reviewers freeze up when it comes to describing the sound of a speaker and become amazingly conservative—the speaker has, for example, a "recessive" midrange, whereas the midrange of the amplifier just reviewed is "chocolaty" or "dark-hued," right? It would appear that reality is more inhibitive to colorful description than fantasy.

I also want to reiterate here my great reluctance to publish the printouts of my measurements, although I keep getting requests to do so. I have a great fear of misinterpretation by the technically semiliterate, which could then be straightened out only by publishing even more printouts, resulting in page after page of little or no value to the average reader. I believe that a basic truth is better expressed in a few well-chosen words than a picture, except in certain rare instances when, of course, I'll make an exception and publish the picture. I further believe that the plethora of "scientific" graphs in certain high-end journals is a cosmetic cover-up for the lack of science in their basic approach to equipment evaluation.

Who wrote which review?

Two of the reviews that follow are by David Rich, who loves small speakers, whereas I find them a source of frustration in my large listening room. His byline appears at the head of those two reviews; the other seven are my humble handiwork. All of the measurements discussed by either one of us were taken in my laboratory.

ACI *(Audio Concepts, Inc.)* G3

Audio Concepts, Inc., 901 South 4th Street, La Crosse, WI 54601. G3 floor-standing 3-way loudspeaker system, with base, $829.00 the pair (direct from ACI, fully assembled, including shipping charges). Full kit (without base), $709.00 the pair (including shipping charges). Tested samples on loan from manufacturer.

ACI is how Audio Concepts, Inc., would now like to be known. As longtime readers know, I look with disfavor on all forms of alphabet soup, of which we have far too much in audio (as we do of companies having names starting with *audio,* for that matter). Hey, what's wrong with Mike Dzurko, Inc., named after the owner? Isn't that more distinctive? Mike has a great deal to be proud of, but perhaps he is too modest to tout his own name. Or maybe he thinks it sounds too Slavic—but did that hurt Stravinsky in the music business? Anyway, ACI it is.

The G3 is one of the things Mike can be proud of because for the price of so many other mediocre, not-quite-full-range speakers this model offers you a 40 Hz to 20 kHz system without significant faults, good enough in just about every important respect to satisfy the discriminating audiophile. That's an achievement not to be sneezed at. If you opt for the full kit version, the value per dollar becomes even more remarkable, since relatively little work is involved.

The speaker consists of a 10" woofer in an "aperiodic" enclosure vented rearward through five $^{9}/_{16}$" holes, a 5" midrange driver in an isolated subenclosure with a single 1" hole in the back, and a 1" ferrofiuid-damped aluminum-dome tweeter firing through a flat doughnut of felt. The whole structure stands about 40" high, which includes the 4" high screw-on base. The width and depth are about a foot each. The cabinet is solidly constructed of ¾" stock and finished in wood veneer (several choices available) on the sides and top only. The black knit grille cloth is stretched on a beveled frame, not as ingeniously antidiffractive as that of the Thiel CS2.2, for example, but showing some attention to the diffraction problem nonetheless.

The impedance characteristics of the speaker present an easy load to the amplifier. Above 100 Hz or so, past the box-tuning convolutions, the magnitude stays between 4.2 and 10 ohms, the phase within ±22.5°. Can't ask for anything simpler.

The bass response of the G3 goes down to 40 Hz, the approximate -3 dB point, and declines at the rate of 12 dB per octave below that. Don't confuse such a profile with that of a vented box tuned to 40 Hz; the latter would have considerably less output in the 20 to 30 Hz region. The G3 delivers genuine low bass without boom; the aperiodic tuning assures good damping. (Yes, it's a "fast" woofer in untutored tweako terminology.) The woofer is at the bottom of the enclosure, near floor level—to avoid the "floor bounce" identified and demonized by Roy Allison—leaving about 13 inches (meter) of land between woofer and midrange, which made it a bit difficult for me to measure the 1-meter frequency response of the system between 200 and 1200 Hz. The crossover to the midrange is in that region (350 Hz, 12 dB per octave); above 1200 Hz my 1-meter MLS measurements with the microphone aimed at the midrange/tweeter boundary should be completely trustworthy, reading ±2.5 dB on axis, all the way up to 20 kHz. The response is actually ±0.75 dB between 10 and 20 kHz! That obviously good tweeter (made by Vifa) is crossed over at only 6 dB per octave; fortunately it comes in quite high, above 4 kHz, so that the lack of bottom-end filtering creates no major power-handling problem. Off axis I measured some significant dips in the vicinity of that 4-kHz-ish crossover, of the order of 5 dB at 30° off horizontally and 15 dB at 30° off vertically. That's not at all surprising with a first-order crossover.

In the time domain I observed no anomalies. A positive-going pulse makes all three drivers push forward; square pulses of any width cannot be coherently reproduced, but then I've never seen it done in a 3-way system regardless of the crossover; and neither pulses nor tone bursts revealed any storage patterns. It's a clean machine (to quote Paul McCartney).

In general I discern no grand design behind the driver placement, enclosure geometry, and crossover configuration of the G3, just good, pragmatic decisions based on results vs. cost. The drivers are good, the overall construction is good, the crossover is good (at least on axis)—so the sound is good. Indeed, the sound is more than just good; it's quite remarkable. I never listen to speakers in this price range for more than just a few hours, long enough to form an opinion, but this one I left in my system for five days and remained basically happy, although I'm accustomed to the sound of much costlier speakers. The sound of the G3 is open, smooth, finely detailed, and quite precise in imaging. I prefer it to the sound of the Sapphire II*ti* (see below), which in its current incarnation has a much less pleasant top end and, of course, needs a subwoofer. If I had to find fault with the G3—as I'm not really inclined to—I'd say it sounds just a little too smooth and rounded in the treble, lacking sufficient bite (as distinct from edginess), probably as a result of those off-axis suckouts in the most sensitive range of the ear, where we naturally expect to be overstimulated rather than spared. It's not a serious objection; at least the sopranos never scream at you. For the money, and then some, this is an outstanding speaker.

ACI Sapphire II*ti*

(Reviewed by David Rich)

Audio Concepts, Inc., 901 South 4th Street, La Crosse, WI 54601. Sapphire IIti compact 2-way loudspeaker system, $964.00 the pair (direct from ACI, fully assembled, including *shipping charges). Full kit, $884.00 the pair (including shipping charges). Tested samples owned by the reviewer.*

This speaker is a modification of the Audio Concepts Sapphire II minimonitor reviewed in Issue No. 16. The modification involves the replacement of the modified Focal T120KT Kevlar inverted-dome tweeter with a modified Focal T120Ti, which has an inverted dome made of titanium. An additional modification to the speaker is that its price has been increased to $964.00 the pair. It is also available as a kit for $80.00 less. Should you decide to return the assembled speaker—remember, the speaker is only sold direct—after the at-home audition period (which has been reduced from 30 to 15 days), ACI will no longer refund the assembly charge.

The tweeter modification gives us a chance to reassess the speaker, this time using the Audio Precision MLS test setup to measure frequency response. The new tweeter appears to have rearranged the large resonant peak of the original tweeter at 17 kHz; in one of my samples there is now a 4 to 5 dB peak at 14 kHz followed by a comparable dip at 17 kHz; in the other sample the peak is fairly well suppressed but the dip is not. Tone burst testing showed significant energy-storage effects above 12 kHz in both samples. This is hardly the performance one expects from a state-of-the-art tweeter design; indeed, many soft-dome tweeters perform better. The woofer also showed significant energy-storage effects on tone burst tests above 2 kHz.

The titanium tweeter may be more efficient than the old Kevlar unit, since the on-axis MLS frequency response curves of the II*ti* indicate that the average tweeter level is 1 or 2 dB above the woofer level when the microphone is aimed at the apex of the woofer. A 5 to 8 dB dip (depending on the sample and the microphone position) in the amplitude response between 2 and 3 kHz is the only other significant error in the frequency response curve. Frequency response runs at 30° off axis show less difference in level between woofer and tweeter but an additional dip of 5 dB between 4 and 5 kHz appears in the curve. Whether the grille is on or off the speaker does not significantly affect the amplitude response. The 30° off-axis measurements also show a phase variation of less than 45° from 300 Hz to 15 kHz (phase variation is greater on axis), justifying the claim that this is close to a minimum-phase design. But a large price is paid for minimum phase—the drivers exhibit very significant interference effects when the speaker is used above or below its optimum horizontal plane. (See Issue No. 16 for more details on the cause of this effect.) Aiming the microphone at the apex of the tweeter resulted in an amplitude dip between 4 and 7 kHz with a minimum value of 16 dB. Clearly, great care must be taken when these speakers are set up to insure the woofer is at ear level. The two samples were typically matched within 1 dB, but variations of more than 2 dB were observed in some frequency ranges.

Before the tweeter update the Sapphire II was the principal speaker in my reference system. The unmodified speaker, while lacking the transparency of high-priced state-of-the-art speakers, gave an excellent account of itself over a wide range of program material. I used the equalization function of the Cello Palette (more on this preamp-equalizer will follow if a loan can arranged from the manufacturer) to verify that the transparency loss is partially due to the amplitude dip in the 2 to 3 kHz region. I had my Sapphire II speakers updated to the II*ti* by ACI, so a direct comparison between the new and old tweeters was not possible. From my memory of the unmodified Sapphire II, it appears the new tweeter has brightened the sound of the speaker and the treble range is more colored. The speaker is now less forgiving of problems in program material, but the reproduction of audiophile-quality recordings has also been degraded.

Given the speaker's price increase and performance decrease, the Sapphire II is no longer the remarkable bargain it once was. The speaker would still be good enough to warrant a risk-free audition, but the audition is not risk-free under ACI's new policy. The cost of shipping and nonrefundable fees add up to over $115.00. The speaker is, in my opinion, not good enough to justify taking this chance. I can thus only recommend audition of the speaker in its kit form. Assembly of the kit requires significant care to prevent damage to the fragile drivers. The speaker costs $80.00 less as a kit, which reduces the risk of return to the $35.00 shipping charge and the time required to assemble the speaker.

*[Having not only measured but also listened to David Rich's modified speakers in my laboratory, I agree with his opinion about the decline of this product. The original Sapphire was an awkwardly packaged, somewhat impractical but marvelous speaker that gave the Quad ESL-63 a hard time in side-by-side comparison. The Sapphire II was almost as good and a much more sensible package. The Sapphire II*ti *sounds slightly wiry and sibilant—a very untypical quality in an ACI speaker—and lacks the sonic refinement and class of the original design. Maybe Mike Dzurko listened to some not-so-good advice instead of his own good ears.—Ed.]*

MACH 1 Acoustics DM-10

MACH 1 Acoustics, RR 2, Box 334A, Wilton, NH 03086. DM-10 floor-standing 3-way loudspeaker system, with accessory granite base and grille, $7995.00 the pair. Tested samples on loan from manufacturer.

Let me say it before we get involved in the details: this is one of the finest loudspeakers known to me, regardless of price. Like everything else in this world, it has its limitations, but those limitations are intrinsic to the basic concept and intended purpose of the speaker; they aren't design faults. The speaker is intended for extremely refined, high-resolution playback at not excessively high levels in not excessively large spaces, and it accomplishes that faultlessly.

The key to the design is the choice of drivers. Marc McCalmont, the designer of the DM-10 (he is a Marine flier turned Pan American pilot turned audio entrepreneur), chose the Accuton 1" inverted-dome tweeter and 3½" inverted-dome midrange, and a 9½" Dynaudio woofer. The Accutons have ceramic diaphragms made by vapor deposition and are billed to the manufacturer at approximately $160 and $200, respectively. They are quite fragile and need to be crossed over just so to keep them out of trouble. The Dynaudio is also ridiculously expensive, so that Marc pays over $1000 up front for drivers before he has even started to put other parts into a pair of speakers. Welcome to the world of High End. I must say, however, that these are better drivers than you get in, say, a Wilson WATT.

The cabinet of the DM-10 has 1" walls, except the front bafflle, which is made of 1¾" damped laminate. This guy doesn't fool around. The dimensions of the box are 44" high by 11" wide by 14½" deep; the front edges are rounded; the finish is in your choice of veneers; the grille is optional, the basic design having been conceived with fully exposed drivers. The woofer is located only a few inches above floor level to avoid "floor bounce" (see the ACI G3 review above); the midrange and tweeter sit high and are offset inboard, resulting in a mirror-image pair. The woofer is in a sealed enclosure; the crossover slopes are fourth-order (24 dB per octave); the network is made with air-core inductors (except in the woofer circuit) and polypropylene capacitors; the crossover frequencies are approximately 250 Hz and 3 kHz. The general design philosophy is to be textbook correct and never mind the cost. No tricks, no surprises, no compromises.

I found only two basic design characteristics that I —putting myself in the place of a purchaser—would have wished to see improved at this exalted price level. One is the bass, which is very clean and well-controlled but could go deeper in a box of this size. (An off-the-shelf woofer, no matter how costly and how magnificently made, hardly ever has the exact Thiele-Small specs for the particular system optimization one needs.) The nearfield response I measured was flat down to an f_3 of 44 Hz and declined 12 dB per octave below that—a classsic sealed-box profile. The impedance curve indicates that the box is tuned to 35 Hz, so the system must be slightly overdamped. As the nearfield response at 30 Hz is only 10 dB down, the "room gain" in smallish rooms should bring it up a tad, but in my big room I would have preferred stronger bass. The other small weakness of the design is that the little Accuton tweeter is somewhat deficient in power handling, so that you have to watch the level in opera recordings, for example, because the soprano's fortissimo high notes tend to sound a bit strained if you turn up the volume. This is a medium-signal, rather than a large-signal, transducer.

That said, I must then immediately add that at normal to moderately high levels the sound of the DM-10 is exquisitely beautiful and transparent, absolutely world-class. Both texture and structure—to use the John Eargle terminology which is so superior to the high-end tweako vocabulary—are as accurately reproduced as anyone could wish for. Furthermore, the crossover design and driver mounting/positioning are such that the speaker isn't the least bit temperamental when it comes to placement—the soundstage doesn't collapse and the balance doesn't go to hell when you move the cabinets eight inches this way or that way. (Marc McCalmont has written an entire manual on room acoustics and speaker placement, by the way.)

In measuring the 1-meter response of the speaker with the MLS technique, I didn't run into the same problem as I did with the ACI G3, although the vertical distance between woofer and midrange is even greater in the case of the DM-10. The much steeper crossover slopes are probably the reason. On the tweeter axis, the response was ±3 dB from 300 Hz to 20 kHz, which is even better than you'd think because in the crucial three octaves from 1 kHz to 8 kHz the deviation from absolute flatness was only ±1.25 dB. And that's not all. There's hardly any change in the response up to 10 kHz at 30° off axis; only the 10 to 20 kHz response starts to slope downward a bit. No wonder the speaker sounds great.

In the time domain, I observed nothing that could change my high opinion of the DM-10. Pulse coherence was of course nil; it's a spread-out 3-way system with high-order crossovers to begin with, and a positive-going pulse pushes the tweeter diaphragm inward, whereas the midrange and woofer diaphragm move outward. It's academic; the proof of the pudding is in the superior frequency response on and off axis. I did see just a tiny bit of garbage between tone-burst envelopes but not enough to attribute any importance to.

The impedance characteristics of the speaker indicate the need for an amplifier of good but not exceptional current capability; above the impedance swings due the box, the magnitude stays between 3 and 8 ohms and the phase within ±25°. Efficiency is of the order of 87 dB, which is about average for speakers in this format.

Where do I rank the MACH 1 Acoustics DM-10? If sheer transparency, refinement, and naturalness of sound are the top priorities, it ranks very close to the top. I haven't tested everything, of course, but its only competition known to me in that super-finesse category is the Win SM-10. If, on the other hand, the big sound, life-size dynamics, deep bass, and generally awesome impact are the desired traits, then it lags behind the Waveform Mach 7, the Carver "Amazing Loudspeaker" Platinum Mark IV, and others of that ilk not yet reviewed, which are slightly cruder in sonic texture, at least in my opinion. In any event, although the DM-10 is fairly priced considering the manufacturer's cost of parts and labor and the dealer's normal markup, it's still a classic case of "if you have to ask the price you can't afford it."

Monitor Audio Studio 10

(Reviewed by David Rich)

Monitor Audio Loudspeakers, Kevro International, Inc., P.O. Box 1355, Buffalo, NY 14205. Studio 10 compact 2-way loudspeaker system, 2549.00 the pair (without stands). Tested samples on loan from distributor.

Editor's Note: *This review had already been written and was moving slowly through our editorial pipeline when Monitor Audio announced their new Studio 6, replacing the Studio 10. We already have the new model on hand as this issue goes to press, and the differences appear to be minor, certainly not significant enough to require the withdrawal of this review. The Studio 6 introduces some improvements, which will be the subject of a briefer review in Issue No. 20, using this review as background.*

* * *

Monitor Audio is a small British speaker manufacturer. A distinct advantage of Monitor Audio over small American manufacturers is that the company custom designs its own transducers. The Studio 10 has custom-designed drivers exclusive to this unit. Monitor Audio was one of the first manufacturers to use metal-cone driver technology. In the Studio 10 both the woofer and tweeter diaphragms are made of metal. The use of an aluminum/magnesium alloy in the tweeter dome and a proprietary anodizing process are claimed to reduce tweeter resonance. Ceramic coatings and a three-stage metal drawing process for stress release are claimed to reduce cone resonance in the woofer. The large R & D expense involved in developing the drivers and the associated manufacturing process is reflected in the price of the Studio 10. What is unacceptable to me at the speaker's price point is the fact that the five-way binding posts at the input of the speaker are not 0.75" spaced, so you cannot use double banana plugs. *[Britannia no longer rules the waves in high-end audio but she still waives the rules. Actually, David, old boy, the measure used to determine the distance between those ruddy binding posts, don't you know, is King Henry VIII's thumb.—Ed.]*

The cabinet is made of ¾" thick Medite medium-density fiberboard. The speaker's woodwork is of the quality found in high-end furniture. As an example, the cabinets are veneered inside the box as well as outside. This prevents the wood from warping by balancing the stress on the wood exerted by the veneer. For $500.00 more a rosewood piano finish is available. I would suggest the availability of a lower-cost vinyl walnut version (that way the speakers would match the rest of the furniture in my apartment), but the master wood craftsman at Monitor Audio would almost certainly be insulted by my even suggesting such an idea. The structural rigidity of the cabinet is more important to a speaker's performance than the cabinet's looks. The cabinet lacks the extensive crossbracing of the ACI Sapphire II and appears (using the high-tech knuckle-rap test) to be more resonant. D. B. Keele Jr. in his review of the speaker in the July 1991 issue of *Audio* found a slight cabinet resonance at 445 Hz, using a more sophisticated test metrology. Perhaps to minimize the effect of the resonance, a pair of optional lead-filled metal speaker stands, weighing over 65 pounds each, is available. Any small improvement in the sound quality of the speaker is outweighed by the speaker stands' $850/pair price. In addition, the use of these stands cancels out one of the chief advantages of a mini-monitor—the ability to move the speaker easily. Another problem with these stands is that the height of the speakers above the floor is not adjustable. When a loudspeaker uses low-order crossovers (second order in the case of the Studio 10) and is auditioned at relatively close distances (as would be the case in the small rooms this speaker is designed to be used in), large dips can occur in the response if the height is not adjusted optimally.

Except for a broad peak between 3 and 7 kHz, the anechoic frequency response of this speaker fits in a ±2 dB window from 300 Hz (the low-frequency limit of the test) to 20 kHz. This measurement was made with the microphone aimed between woofer and tweeter. The peak, which is due to the resonant behavior of the large "phase plug" of the woofer, has a maximum value of 5 dB at 5.5 kHz. The peak was almost identical for both samples of the speaker. Tone burst testing showed energy storage problems in the woofer's phase plug near the 5.5 kHz peak, but the woofer's performance was exemplary below the peak's frequency. The tweeter showed no energy storage problems at all below 20kHz, an excellent result. The tweeter's resonance occurs at 26 kHz. This is almost an octave above the resonance measured in the Focal tweeter used in the ACI Sapphire II*ti*. Matching between the samples was superb, as it was held to within 1 dB.

Changes in amplitude response near the crossover region are significant but much less severe than with the ACI Sapphire II*ti* in vertical off-axis tests, and the changes are symmetrical. This is indicative of the drivers' being inphase at the crossover point. An even order (2nd, 4th, 6th...) crossover will drive the speakers inphase at the crossover, but Monitor Audio claims that the electrical crossover is 6 dB per octave, which would result in the drivers' being 90° out of phase at the crossover. The discrepancy, as explained by Monitor Audio design engineers, occurs because the mechanical rolloffs of the drivers are designed to exhibit a 6 dB per octave slope in the crossover region. The sum of the mechanical and electrical rolloffs yields a 2nd-order crossover response. Custom designing the drivers' rolloffs is possible only because Monitor Audio does its driver design in-house.

Our nearfield measurements of the woofer and the port showed the bass response of the speaker to be flat to 60 Hz (the -3 dB point); then it falls at 24 dB per octave. Given the small size of the woofer and the cabinet, distortion in the bass region becomes high as drive level is increased. Sonically this translates into the complete loss of the bottom octave of the orchestra, and the octave above that is reproduced with less definition than by larger speakers costing far less than the Studio 10. Sub woofer users be warned: it is very difficult to mate a 4th-order vented box to a subwoofer without incurring significant amplitude variations near the crossover region.

With the speakers placed on the optimal vertical axis they are slightly forward-sounding but much less so than would be expected given the 5.5 kHz peak measured in the speakers' amplitude response. The explanation may be found in the room response measurements made by D. B. Keele in his review of the speaker. The amplitude of the peak appears to be reduced in the room response curve. These speakers produced a remarkably transparent sound. In comparison with the ACI Sapphire II*ti*, instrumental timbres were reproduced with much greater clarity and inner voices were more easily distinguished. Dynamic contrasts were enhanced by the Studio 10. The audible differences were significant enough to justify the price difference between the two speakers.

At first I thought the transparency of the Studio 10s would make less than optimally recorded CDs unlistenable, but just the opposite occurred. With less distortion coming from the loudspeaker, many of these recordings proved to be more listenable than before. The ability of this speaker to bring out the best in old CBS, Philips, and RCA remasters was its most remarkable quality. Of course, nothing could be done to make truly bad recordings (such as most Deutsche Grammophon digital CDs) listenable. On modern source material these speakers produced the best sound I have experienced in my listening room. Thanks to the speakers' small size and careful driver matching, imaging was excellent. In your Editor's listening room the speakers were bested by the MACH 1 and Win loudspeakers, but both of these speakers cost more than twice as much as the Studio 10s. The principal difference was the elimination of the forward-sounding character of the Studio 10s (the sophisticated diaphragm materials used by MACH 1 and Win do not show the resonance effects of metal cones) and a more extended and less distorted bottom end. To my ears the Studio 10s were more open and transparent than the Waveform speakers, but the Waveforms can play much louder and have subwoofer-quality bass.

A favorite trick of high-end dealers is to claim that a speaker of the quality of the Studio 10 can only be driven with electronics costing five figures. Do not believe this. The Studio 10s are easy to drive. The impedance never goes below 5.5 ohms and the phase angle varies less than ±30 degrees. I temporarily replaced my current electronics with my 15-year old Audire Legato/Crescendo electronics and heard virtually no degradation in the Studio 10s' performance. The combination of the Studio 10s and a good-quality mass-market integrated amplifier and CD player could cost less than $3200. The difference in sound quality between this system and a similarly priced high-end dealer system would be almost comical. If you have a small room, listen at sound levels that will not break a lease, and do not require extended bass response, the Monitor Audio Studio 10s are highly recommended.

[In my large listening room, connected to my Boulder electronics, the Studio 10 didn't delight me nearly as much as it did David, although I do not disagree with his point-by-point analysis of the speaker's characteristics. The 5.5 kHz peak and ringing were obviously more offensive to my ears than his, but I happen to be very touchy in the 3 to 6 kHz octave, more so than most people. In any event, the woofer of the new Studio 6 has no comparable phase plug, so that may turn out to be a dead issue.—Ed.]

Tannoy 615

Tannoy Ltd, c/o TGI North America, Inc., 300 Gage Avenue, Unit 1, Kitchener, Ont., Canada N2M 2C8. Model 615 floor-standing 3-way loudspeaker system, $1599.00 the pair. Model 6s1 base, $99.00 the pair. Tested samples on loan from distributor.

Tannoy is one of the oldest English brands in loudspeakers, and Model 615 is the top of the Tannoy "Sixes" line. All models in the line are six-sided columns of various heights—the cross section is a kind of chopped-off hexagon—and the costlier models feature drivers in Tannoy's classic Dual Concentric format. The 615 has an 8" Dual Concentric main woofer/tweeter, an additional 8" woofer, and an 8" passive radiator, each with a separate snap-on, floating-type grille. In the main driver, a compression-type high-frequency transducer is horn-loaded by the specially contoured woofer magnet and woofer cone. All three 8" cones are made of a very inert high-tech plastic material.

The remarkable thing about the 615 is that, while its footprint is slightly smaller in area than this page and its height is only a couple of inches over three feet, it sounds like a big speaker! It has some colorations, to be sure, but these are of the brassy, pro-sound variety, not the closed-down, nasal, unmusical kind exhibited by unsuccessful audiophile-oriented designs. I like this speaker, despite its small shortcomings, and the younger generation in my house also preferred it in some ways to the more puristic, neutral-sounding Thiel CS2.2, for example, because of its dynamics and impact. It's definitely not a wimpy speaker.

One reason is that the 615 is very efficient; its 1-watt/1-meter SPL rating was a little hard for me to nail down because of its wide swings in impedance, but the figure is well above 90 dB and could be close to 92 dB. About that impedance—once above the range of the tuned box, the magnitude rises from 3 ohms to 28 ohms over five octaves, then falls back to 5 ohms over two octaves. The phase also swings widely within ±45°. Not exactly a resistorlike load for the amplifier, but the good ones can handle it with aplomb. The box with its passive radiator is tuned to 29 Hz; maximum passive radiator output is at 25 Hz; the nearfield summed response is -3 dB at 35 Hz and within ±2 dB up to 200 Hz. That's remarkable bass response for a small, efficient speaker system in which the woofer is fairly well level-matched to the midrange. The low bass doesn't hold up too well at very high levels, but what did you expect—this is a very compact floor-standing speaker. Power handling above the bottom two octaves is, on the other hand, truly excellent.

The frequency response from 300 Hz up is a little humpy and notchy but still fits into a ±3 dB strip up to about 16 kHz; I've seen better and I've seen worse. At least nothing sticks out terribly. Off axis the picture is very nice; up to almost 10 kHz you have to go more than 45° off axis to see a serious rolloff, and the pattern is of course symmetrical up and down and sideways because of the coaxial wave launch. It should also be noted that the first-order electrical crossover at approximately 2 kHz between the two coaxial elements doesn't come with the usual penalties (lobes, tweeter distortion, etc.), thanks to the coaxial geometry and the power handling of the compression driver. The crossover network also rolls off the separate woofer above 400 Hz; the two woofers are in parallel, but only that of the coaxial unit is active up to the tweeter crossover. (The passive radiator isn't connected to anything; it acts as a vent.)

My time domain measurements indicated that the two woofer cones move outward in response to a positive-going pulse, but the tweeter diaphragm moves inward. The tweeter is wired with reverse polarity, perhaps to compensate for the total second-order profile resulting from the acoustical rolloff added to the first-order electrical slope. In any event, the resulting pulse coherence is quite good, though far from perfect; tone bursts reveal no evidence of storage but plenty of interference between the coaxial drivers, which of course are not coplanar. That may be one reason for the slight colorations heard, in addition to the not quite smooth frequency response. Imaging, however, is very good, probably because of the symmetry of the radiation pattern.

One practical feature of the 615 is its fake-marble top, perfect for your drink while you change CDs; an impractical feature is—here we go again, God save the Queen and pass the crumpets—the Britannic spacing of the binding posts at the input, making it impossible to use double banana plugs. I would gladly trade the 615's two pairs of inputs—a genuflection to the tweako doctrine of biwiring—for one pair with ¾" spacing (believe me, blokes). Even so, this is a special speaker with a special flavor, and after getting my reservations off my chest I'm quite willing to endorse it. After all, how many waist-high speakers with a small footprint have large-signal capability, decent bass, and outstanding dispersion?

Thiel CS2.2

Thiel, 1026 Nandino Boulevard, Lexington, KY 40511. CS2.2 floor-standing 3-way loudspeaker system, $2250.00 the pair. Tested samples on loan from manufacturer.

According to a PR release received in mid-December, the Thiel CS2.2 will henceforth be known as the CS2 2, prounounced "two-two." Huh? What's that? Yeah, a space in place of the decimal point. You see, Bose Corporation owns the trademark to the number 2.2; they currently sell a speaker called a 2.2 II. The press release didn't say that Bose demanded the change, but to me it sounds typical of their corporate mentality. Well, I'm not *selling* anything called 2.2; I'm an editor in a country that guarantees freedom of the press and I can call the Thiel speaker Irving, Attila, or even Amar if I want to; so for the purposes of this review I'll just stick to the old CS2.2 designation. (Whew, that was a close one, Amar, wasn't it?)

I also want to say, before anything else, that I have the greatest respect for the Thiel company as an audio manufacturer, for Jim Thiel as a speaker designer, and for Thiel speakers as products of absolute integrity—but also that I disagree with the Thiel philosophy of loudspeaker design on a very fundamental level. I admire the product, but it isn't what I want for myself as a music lover and audiophile. I fully understand, however, those who are of the opposite opinion.

Now then, the Thiel CS2.2 is a 3½-foot high speaker with a footprint of just over a square foot and a slanting front. Every detail of its design reflects a single-minded pursuit of quality, as if the designer had never heard of money-saving solutions, not even in this low-end-of-the-high-end price range. The front baffle is 2" thick, the cabinet walls 1" thick, the bracing truly massive. The back is finished just as beautifully as the sides, and the specially contoured baffle fits deep into hollow of the specially beveled grille to form the neatest, most ingenious nondiffractive front of any speaker I've seen—I would have died for such a design in my speaker manufacturer days. Every inch of the speaker shows serious thought and loving care. The driver complement consists of an 8" woofer with a novel two-layer diaphragm, a 3" midrange driver, and a 1" metal-dome tweeter. The woofer is in a tuned enclosure using a 6" by 9" elliptical passive radiator. The crossover network is the heart of the design, consisting of 26 circuit elements and built with a total of 35 components, including the highest-quality air-core coils, polypropylene and polystyrene capacitors—nothing but the best. Its purpose is to synthesize perfect first-order acoustical crossovers and achieve perfect phase and time coherence. That is Jim Thiel's obsession, for which he is willing to trade off many other important performance characteristics, and that is where I part company with his design philosophy.

On the axis of the midrange driver, the CS2.2 has the most perfect frequency response of any speaker I have tested so far. The 1-meter MLS measurements show the amplitude response to be within ±1 dB from 1.5 kHz to 19 kHz and ±2.5 dB from 300 Hz to 20 kHz; the phase response stays within +15° from 300 Hz to 20 kHz. I suspect that even better results are obtainable at 2 or 3 meters because the center-to-center distance between the midrange unit (which is crossed over at 800 Hz and 3 kHz) and the woofer (which takes over below 800 Hz) is about meter, i.e., a significant part of 1 meter. The off-axis response at a 30° horizontal angle from the midrange axis is also spectacularly good, almost as flat as the on-axis response, just 4 or 5 dB lower in level. But—a very big but!—move the microphone up or down vertically, away from the "sweet spot," and all bets are off. The response goes to hell. Welcome to the world of first-order crossovers. (Siegfried Linkwitz didn't spend all that time and effort perfecting fourth-order crossovers for nothing.) In time domain tests, square pulses of various durations confirm the claims of outstanding coherence at the sweet spot—and only there.

And that's not all. At fairly moderate levels, sine waves in the 300 to 400 Hz range, a full three octaves below the tweeter's nominal 3 kHz crossover, make the tweeter buzz! Why? Because the first-order highpass filter is only 18 dB down in that range, and that's apparently not enough to protect the tweeter from out-of-band overload. Musical program material containing a fair amount of 300 to 400 Hz energy will do the same thing; luckily it doesn't happen very often. (No, neither tweeter was defective; they sounded very clean in their working range. It's possible that Vifa, their manufacturer, doesn't consider buzzing to be an issue because the higher-order crossovers used in most speaker sytems provide adequate protection from low-frequency excitation.)

Regardless of the tweeter buzz, the overall power handling of the CS2.2 is unimpressive. The bottom octave isn't reproduced cleanly at high signal levels, although the small-signal bass response is very good—I measured -3 dB at 32 Hz and essentially flat response above that. When the music gets really loud over a wide frequency range, as in a Mahlerian climax, the little 3" midrange driver begins to sound mildly distressed, its nominal two-octave passband being in reality more like six octaves with those first-order rolloffs at either end. It's a case of cruelty to small creatures.

Part of the power-handling problem is the low efficiency, necessitating more than the usual amount of drive for realistic sound levels. The standard input of 2.83 V produces an SPL of 86 dB at 1 meter, but when you look a little closer that's actually 2 watts because the nominal impedance of the speaker is 4 ohms. So the 1-watt/1-meter efficiency rating is in reality more like 83 dB. On the other hand, the load presented by the CS2.2 is a piece of cake for almost any amplifier to drive. The magnitude of the impedance is almost flat, hugging the 4-ohm line and staying between 3.3 and 5 ohms at all frequencies above 100 Hz. The impedance peaks of the tuned box are only 7 ohms. As for the phase of the impedance, it stays within ±10° at all but the lowest frequencies. Jim Thiel loves flat response curves.

Having said all that, I must not fail to emphasize that the sound of the CS2.2 at moderate levels—and with the listener's ears within a certain window, not too high and not too low—is outstandingly good. Nothing sticks out, everything is wonderfully smooth and neutral, and the imaging is nothing short of superb, as good as I have heard in my setup. I know that Jim Thiel attributes that subjective perception to the Miracle of Saint Coherence, but I don't agree. I suggest the following experiment, which is beyond the purview of an audio journal but not of a speaker designer:

Build three versions of the CS2.2, or of any other comparable Thiel model for that matter. No. 1 would be the standard factory version. No. 2 would be the standard version with a well-designed fourth-order Linkwitz-Riley crossover substituted for the first-order Thiel crossover. No. 3 would be the standard version with the first-order crossover but in a square-cornered cabinet with sharp edges, no bevels, no contoured front baffle—just a regular "monkey coffin." I'm willing to bet the ranch that No. 2 would image just as well as No. 1 (and, incidentally, have better power handling and vertical polar response), whereas No. 3 would image much less well and generally sound a little rougher and less focused. In other words, I contend that Thiel's deservedly prized imaging is due to the superior management of diffraction, reflective surfaces, second arrivals, etc., not to the first-order crossover. I also know that Jim is most unlikely to change his mind on this subject.

In conclusion, I want to mention that long before I got around to measuring and listening to the CS2.2 in my laboratory, I had it hooked up to a more modest but fairly high-powered system in another room for my two sons (both in their twenties) to use as their regular stereo. Their verdict: "It sounds very accurate and uncolored but it doesn't kick ass."

Waveform Mach 7

Ötvös Industries, RR #4, Brighton, Ont., Canada K0K 1H0. Waveform Mach 7 floor-standing 4-way loudspeaker system, $8400.00 the pair, complete with dedicated electronic crossover. Tested samples on loan from manufacturer.

This is not a new speaker system but it has changed sufficiently since my exhaustive and highly favorable review of the original version in Issue No. 14 to warrant a second look. The new Mach 7 designation after the brand name (couldn't John Ötvös have found a more original tag?) indicates the following new features:

The dedicated electronic crossover is now made by Bryston and uses the same basic topology as the superb Bryston 10B general-purpose crossover. The "boost of a little over 3 dB at 16 to 17 kHz to equalize the supertweeter where it begins to roll off (my words in No. 14) has been eliminated. The 1" textile-dome lower tweeter has been replaced by an MB QUART 1" titanium-dome unit. That, of course, necessitated a new passive crossover network for the upper section of the speaker. There is also a new grille incorporating a special felt ring that fits over the dome tweeter. In addition to the incredible virtuoso woodcrafter's custom cabinet (what I called the Bernini version) and the black piano-finished cabinet, there is now a crackle-finished professional-style cabinet—the one I received. The update to Mach 7 is available to owners of the older model for the amazingly small sum of $600.00 (Canadian). John Ötvös is obviously not of the William Z. Johnson school (which holds that mods should not only be offered frequently but also end up being just as profitable as the original sale).

Not that there are many candidates for the update. The speaker, despite its outstanding qualities, just didn't sell—*Stereophile* made sure it wouldn't. (Yes, one more example of a superior audio product railroaded by the Atkinson, Archibald, and Santa Fe. See the review by Larry Archibald and John Atkinson in their November 1989 issue and my comments on the politics of the situation in my original review.) What is truly remarkable is that the endorsement of the Waveform by the professional audio community didn't help. Jack Renner and Michael Bishop at Telarc are using it to monitor their latest and greatest recordings (see the equipment credits in the cover brochures of Telarc CDs). Craig Dory at Dorian has also started to use it. Jack Renner has gone so far as to commission for his own and Mrs. Renner's personal use, in their house, a five-channel surround-sound system using Waveform speakers—in the "Bernini" version! Nevertheless, only 30 pair are currently in the field—such is the influence of the Santa Fe railroaders in the high-end audio market. It didn't help, either, that Harry Pearson had also taken a cheap shot at the speaker in a typically smug, half-assed editorial footnote—not a review—in *The Absolute Sound.* (The review never materialized.) I ask you, friends, who knows more about good sound, and who stands to lose more by being wrong about it—Jack Renner, Michael Bishop, Craig Dory, and their professional crews, or Larry Archibald, John Atkinson, and Harry Pearson? Tough choice, isn't it?

The improvements in the Mach 7 version are quite significant. The frequency response is flatter and smoother. The highs are better balanced with the rest of the spectrum. The difference between on-axis and off-axis response is smaller. Also, the Bryston-made electronic crossover is a more advanced and reliable design, less likely to cause ground loops and more likely to have an unlimited life span. Since the review in Issue No. 14 covered every detail of the design, which is still exactly the same except for the changes mentioned, I have no reason to go over the same ground again. The measurements above the bass range, however, have improved. The on-axis response is now within ±2 dB all the way up to 18 kHz, and the off-axis response up to 45° hews amazingly close to that line, deviating by only 1 to 4 dB under 10 kHz and only 4 to 7 dB in the octave above that, up to 20 kHz. (Yes, I said 45°, not 30°.) That begins to approach the constant-directivity model, and the total radiated power measurements made in the acoustical laboratories of the National Research Council in Ottawa (better than my lab!) paint a corroborative picture. Distortion above the bass range is minuscule, typically -48 dB relative to an SPL of 95 dB (which is unbearably loud as an average), but the bass is slightly more distorted than it would be with a motional-feedback woofer like the Velodyne. That could be the next design update, but the bass as it stands is still pretty awesome, as I originally wrote.

What is the overall impact of the Waveform Mach 7 in terms of subjective listening? I would say that I haven't heard another forward-firing monolithic speaker that equals it as a total package. Its distortionless handling of the most dynamic and complex program material, its deep and unshakably tight bass, its lack of resonant coloration in any frequency band, its extended but neutral top end, its natural tonal balance, its high efficiency, its absolutely clean delineation of whatever music is fed into it—all these virtues together give it the highest composite score in my book. On various individual counts—3-D imaging, see-through inner detail, finesse of texture in the treble, forgivingness in placement—I can name speakers that will beat it, some in this very article. Also, if you prefer planar and/or line-source speakers, the Waveform is unlikely to convert you because it just doesn't launch sound waves the same way. As a sonic decathlon champion, however, I think it's very hard to beat. Paul Barton, the Canadian engineer who designed it for John Ötvös, certainly did his homework.

One small warning. Because of its flat response and wide dispersion in the treble range, the Waveform can sound too bright if highly reflective walls and other bare surfaces are near it. In a very large room, where the walls are far away on all sides, it's not an issue (the delayed second arrivals actually sound nice); in a smaller room wall treatment and careful aiming may be necessary.

I should also add that if your curiosity is aroused now and you would like to check out the Waveform in a dealer's showroom, there are no dealers, alas. To the best of my knowledge, all sales so far have been manufacturer to end user. The bright side of that situation is that the official "retail price" of $8400 the pair is at this point rather theoretical and probably negotiable to some degree in a direct sale. That's just a hint, not a promise.

Westlake Audio BBSM-8VF

Westlake Audio, Manufacturing Group, 2696 Lavery Court, Unit 18, Newbury Park, CA 91320. Model BBSM-8VF floor-standing 3-way "reference monitor" loudspeaker system, $4050 the pair, including pedestals. Tested samples on loan from manufacturer.

Here we have what in my opinion is a highly specialized speaker. For certain users with highly specific requirements it provides the best possible answer—or at least one very good answer. For the general audiophile with four big ones to spend on a pair of speakers there are many other options to explore before taking this route.

What are those specific requirements that make the BBSM-8VF (whew, what a mouthful of alphabet soup!) an attractive choice? Tremendous efficiency, bulletproof power handling, wide dynamic range, low distortion, powerful and well-controlled though not extremely deep bass—and all that, most importantly, in a fairly compact package. If that's what you want, rather than ultraprecise imaging, structural detail, finesse in the treble, gorgeous instrumental textures—the more delicate audiophile priorities—then you'll be a "happy camper" listening to a pair of these Westlake speakers, which have a distinct studio pedigree. Westlake Audio is basically a professional brand; their typical customer is a professional who earns his/her living with audio; their interest in the audiophile market is relatively recent (though currently quite keen). Their audiophile speakers reflect this heritage; for one thing, they're built like the proverbial brick rest room-solid, thick-walled, functional in appearance (oiled walnut, brown grilles, no frills).

To hear the BBSM-8VF at its absolute best, play through it one of the more dynamic recordings by Tom Jung on his *dmp* label. These close-miked, not very reverberant, typical studio recordings of various contemporary bands are generally dominated by hard left, hard center, and hard right information, and have stupendous dynamic peaks. The Westlake speaker eats up that kind of program material and projects it into the room with surgical cleanliness and precision. Complex symphonic material recorded in a large space does less well on the BBSM-8VF; the subtle spatial and timbral cues tend to get lost in a kind of homogenized soundstage, although the climaxes are life-size and undistorted.

The dimensions of this powerhouse are, as I said, quite modest: 31" high, 19¼" wide, 12" deep—something like a large bookshelf speaker doubled. For proper listening height the speaker should be mounted on the matching pedestal supplied, which raises it about a foot off the floor. (My pedestals came with the wrong mounting hardware, by the way. No big deal, just annoying.) The driver complement consists of two 8" woofers mounted side by side in a vented enclosure, a 3½" midrange driver in a separate sealed subenclosure, and a 1" dome tweeter. The crossover network is a somewhat peculiar affair that I had trouble figuring out strictly from my measurements, without a circuit diagram (or taking the speaker apart); it is specified to have crossover frequencies of 600 Hz and 5 kHz, and slopes of 24 dB per octave "minimum," but I'm not so sure after my microphone probings. There seem to be all kinds of slopes. All I know is that the network creates unusually large time displacements; in my time domain measurements a square pulse of 0.5 ms duration was stretched out to 2 ms and one of 1 ms duration to 4 ms. These measurements also indicated that all four drivers move outward in response to a positive-going pulse, but that by itself isn't sufficient for coherence, which in this case is minimal (as in all steeply crossed-over 3-way systems).

The impedance curve of the BBSM-8VF clearly puts it into the specialized category; the magnitude is 4 ohms at 1 kHz but dips well below 2 ohms at 150 Hz and 7.5 kHz, and doesn't exceed 8 ohms even at the vented-box peaks. The phase, above the box range, fluctuates within ±45°. Obviously, the speaker sucks current like an electronic vampire and needs an amplifier with high current capability to feed that habit. Indeed, the whole concept of the design is to draw the maximum power possible out of a good amplifier that doesn't necessarily put out a lot of volts. Westlake even supplies optional speaker cables with plus and minus leads as thick as cocktail wieners and of virtually zero resistance (though highish inductance) just to draw even a tiny bit more current. The speaker has no standard red and black binding posts with banana jacks at its input but comes with a block of screw terminals instead, in order to accommodate the spade-lug terminations of these tweako cables. Terminals are provided for biwiring as well as passive biamplification. (None of that tweaking can *hurt*, of course, but I have little patience with it, as all but our newest readers know, and I refuse to spend time and energy on it until I see scientific proof of its worth.) The right amplifier may be easier to find than you think, however, because the speaker is very efficient, of the order of 90 dB (1-watt/1-meter SPL reading).

The frequency response measurements gave outstanding results. At the best summing junction of the woofers and the vent, the nearfield small-signal bass response was flat down to a -3 dB corner at 25 Hz, closely tracking the box tuning frequency and the maximum-output frequency of the vent. That's quite a bit better than what the spec sheet says. The overall frequency response above the bass range was within ±2 dB up to 12 kHz, with the microphone aimed at the midpoint between midrange and tweeter. A tiny elevation at 15.5 kHz (an extra 1.5 dB or so) prevented that reading from being applicable all the way up to 20 kHz. At 30° off axis horizontally, there was hardly any change up to 10 kHz except a little notch at 3 kHz; between 10 and 20 kHz there was about 5 dB attenuation. These excellent figures explain the basically neutral tonal balance of the speaker.

I think the imaging of the BBSM-8VF would be improved if there were less bare "real estate" on the front baffle and if the edges were rounded or beveled. Westlake will soon be coming out with a new line of speakers called the BBSM VNF series, which will be taller and much narrower, without side-by-side woofers, resulting in a wave launch that should be more to the liking of audiophiles. There will be an almost exact equivalent of this particular speaker in the series. Meanwhile the important thing to remember is that Westlake Audio has its own highly individual and identifiable approach to loudspeaker design, that it's a valid approach, and that you either need exactly that kind of speaker or you don't. I don't.

Win SM-10
(follow-up)

Win Research Group, Inc., 7320 Hollister Avenue, Goleta, CA 93117. SM-10 Broadcast Monitor (2-way coaxial loudspeaker system), $6250 the pair, including stands. Tested samples on loan from manufacturer.

I retested this unique speaker from a new production run that corrected a minor component-mounting error on the circuit board of the original crossover module. This error was claimed to have been the cause of the minor glitch in the crossover region that I had reported. I also wanted to repeat some of my measurements using our newly phased-in MLS technique.

The reader is referred to the original review in Issue No. 17 for a detailed discussion of the design. Here I must, first of all, remedy the failure of that review to specify the impedance of the speaker. The magnitude fluctuates between 3 ohms and 9 ohms; the average is about 6 ohms. The phase fluctuates between +30° and -40°. All in all, a load of fairly limited severity—but not quite a piece of cake—for the amplifier.

The frequency response as originally reported needs no revision in the bass range but has to be qualified further up in the spectrum. All is well on axis up to about 2.5 kHz; I stand by the original +2 dB. Between 4 and 17 kHz the ±2 dB tolerance is again still valid. In that narrow 2.5 to 4 kHz band, however, there is a definite discontinuity, centering on a peak at approximately 3.3 kHz. If that wrinkle is included, the overall frequency response on axis is no better than +3.5 dB from 45 Hz to 20 kHz—a very respectable specification but not amazing. Off axis (measured at 30°) the peak moves down to 2.4 kHz—strange but true!—and the average level of the tweeter drops by 3 to 4 dB, with steeper rolloff above 15 kHz. I am beginning to think that a properly massaged fourth-order Linkwitz-Riley crossover would be better for the Win SM-10 than the variations-on-a-theme-by-Spica network currently used. I also have a feeling that Sao Win is beginning to think so, too. Then again, the whole question may be rendered moot by the dedicated active crossover and amplifier system for the SM-10 he is planning to come out with. All you'll need then will be money.

For the moment, here is the bottom line. Even with its less than perfect passive crossover, the Win SM-10 is the best-sounding loudspeaker known to me in terms of transparency, definition of inner detail, spatial cues, tonal balance, lack of coloration, nonfatiguing top end—in other words, all the finesse criteria. If it could play at very high SPLs and had flat bass down to 20 Hz, it would be the ultimate speaker system, bar none. I think the two main reasons for its superiority are the inherently dead materials used in the construction of the diaphragms and the perfect symmetry of the wave launch. Those appear to be even more important determinants of sound quality than a dB here or a dB there in the amplitude response. (Can't give away too many dB, though.) Anyway, that's just a hypothesis, not a proven fact.

What *is* a fact is that the overwhelming majority of audiophiles who have heard the Win SM-10 would like to own a pair.

Subwoofer
Hsu Research HRSW10

Hsu Research, 20013 Rainbow Way, Cerritos, CA 90701. HRSW10 vented-box 10" subwoofer, $750.00 the pair (with walnut top and standard 12 dB/octave passive crossover). Tested samples on loan from manufacturer.

This very impressive product started its life as the Definitive Research SW10; subsequently discovered and debated name conflicts were resolved by changing both the company name and the model prefix. Dr. Poh Ser Hsu, the scholarly Singaporean technologist responsible for the design, should have had his name on his creations from the beginning; the days when you needed a Waspy name like Lansing or a techie acronym like Altec for your speaker brand are gone forever. We live in the age of Hyundai and Häagen-Dazs. So, in this case, Hsu is the name and deep bass is the game. Very deep bass.

It so happens that I know Poh Ser from his Boston Audio Society days (his tracks stretch from Singapore to the Massachusetts Institute of Technology to the Boston audio mafia to Southern California—a recent move). One of my more frequently quoted *bons mots* is about the Boston Audio Society—in 1977 I wrote that their members "would like to discover an audio Nirvana for forty-nine dollars and ninety-five cents." Poh Ser is very much part of that tradition of penny-pinching audio romanticism, and in his subwoofer design he has brilliantly vindicated it. The HRSW10 is low-frequency Nirvana for $375 per side—one of the two best subwoofers known to me, the other being the Velodyne, comparable models of which range from $1095 to $2750.

How did he do it? The enclosure is a cheap but extremely strong and acoustically inert paper tube with an inside diameter of 14", a wall thickness of ¼", and end pieces made of ¾" fiberboard. This structure is 27¼ " tall and stands on four ordinary ¼" thick hardware-store bolts that raise it 2½" off the floor, for a total height of not quite 30". An inexpensive "sock" made of black knit fabric covers the entire tube except for the top end piece, which has a walnut finish. The downward-facing bottom end piece holds the 10" driver and the 3½" port, which is ducted by a 23" long paper tube. Basically it's a Thiele-Small vented box.

That 10" driver is quite special, however. Since it doesn't have to reproduce any frequencies over 100 Hz or so, it can be totally optimized for low bass. It has a big, heavy 2" voice coil wound in four layers, with sufficient overhang to permit unusually large linear excursion. The cone is made of heavy paper, which is both cheaper and better for the purpose than fancy polypropylene and such. A driver like that, with a properly designed magnet structure, in a ducted enclosure having an internal volume of about 2 cubic feet, can do some fancy woofing. What's more, the relatively thin-walled but rigid tube can't bulge or flex under pressure, even though it weighs practically nothing and isn't braced, because its cross section is a circle and a circle already has the largest possible area for a given circumference, so there's no place for the bulge to go. There are more ways to build a solid woofer cabinet than throwing money at it.

I did my usual nearfield measurements with the driver facing up, a procedure that yields a close approximation to the free-field anechoic response. In the normal position of the subwoofer, the proximity of the floor plus the room gain will of course influence the response, but I wanted to explore the raw input/output capability of the system. At the best summing junction of the driver and the vent, the small-signal response was ±0.5 dB from 20 to 100 Hz! At 10 Hz I measured -6 dB, but that included some contribution to the rolloff by the lab-bench amplifier, which isn't flat to DC. As I kept cranking up the input, the output above 26 Hz remained just as flat; just before I ran out of amplifier power the 20 Hz response was down -2.5 dB, but the 10 to 15 Hz response actually went up. Never, never have I seen anything like it. The distortion may have been slightly higher than with the Velodyne ULD-15 Series II at equal SPLs (I didn't retest the latter side by side), but the Hsu HRSW10 appears to be capable of somewhat higher absolute levels—and who cares about a tiny difference in harmless second-harmonic distortion at 20 Hz? Boston Cheapie is at the very least the silver medalist! Unbelievable.

The vented box is tuned to approximately 15 Hz, where there's a deep null in the output from the driver, and the maximum output of the vent stretches from about 17 to 27 Hz. That's not exactly classic fourth-order Butterworth tuning—I don't know what it is precisely—but the results speak for themselves. The impedance curve in the 10 to 100 Hz range is of course a roller coaster, both in magnitude and in phase (the biggest swings are roughly 110 ohms and ±60°), but the resistive component is 8 ohms.

The subwoofer needs to be biamplified and should be used in pairs (no singles sold and no L + R matrixing recommended). I used a Bryston 4B power amplifier to drive a pair of HRSW10's and a Bryston l0B-sub electronic crossover to match them to various main speakers at different frequencies and with different slopes—but that's traveling first-class and not absolutely necessary (nor in the frugal spirit of the product, for that matter). Just about any old amplifier that can drive 8 ohms will do, and Hsu Research includes with each pair of subwoofers a simple passive network (a second-order RC lowpass filter or third-order if you wish) that goes between the output of the main amp and the input of the sub amp. It's a bit on the Mickey Mouse side in my opinion (I'm not of the Boston school) but it works, and there's really nothing wrong with the concept as long as you're content to drive your main speakers without highpass filtering. For finicky audiophiles there's also a special Hsu Research electronic crossover at a Boston price: $350.00. I haven't tested it. There's some flexibility as to crossover frequency in both the passive and active Hsu crossovers, but you have to specify your needs before you buy.

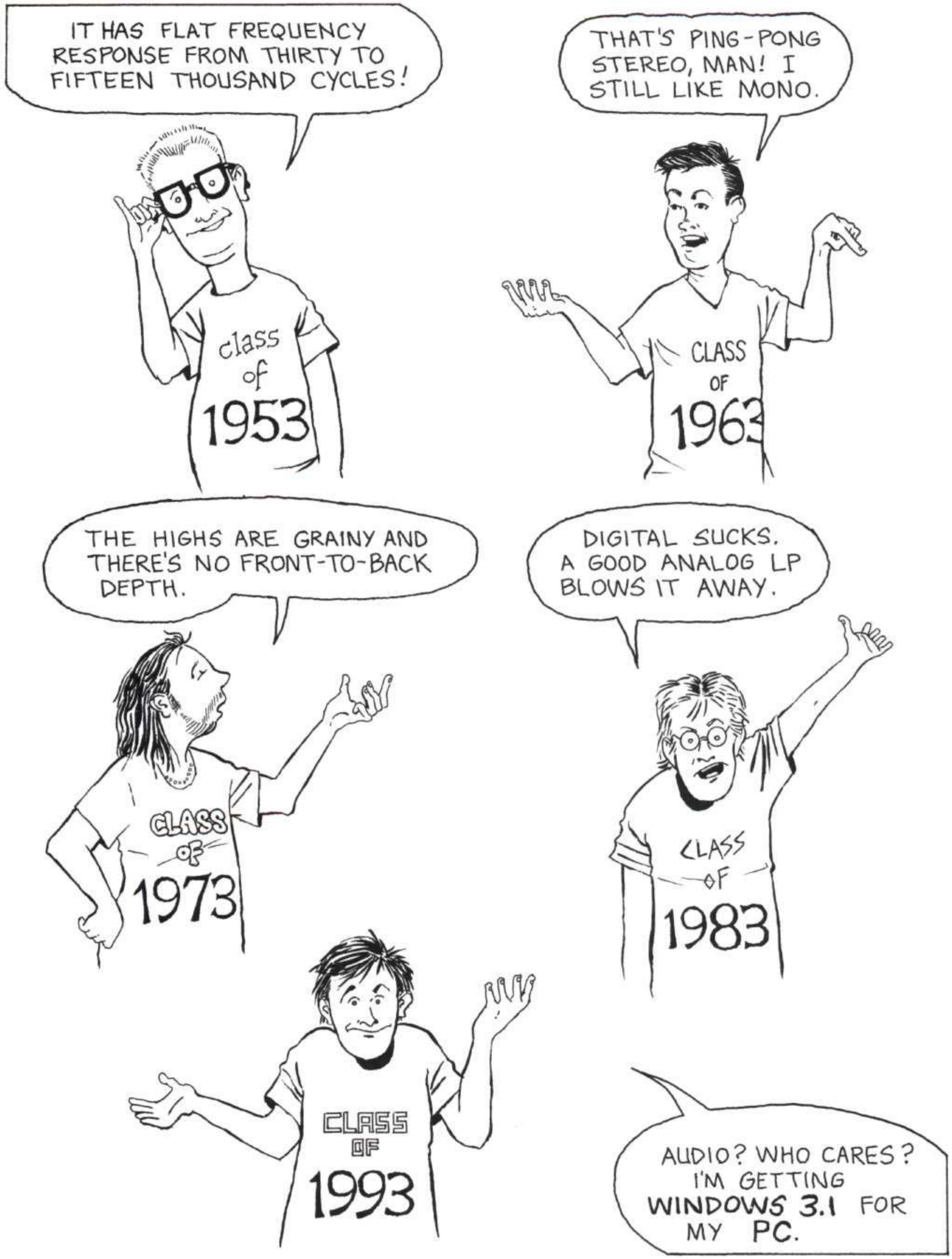
Readers who have been waiting all this time for a quasi-pornographic subjective description of the bottom end obtainable with the HRSW10—how big, how firm, how rumbling, etc.—will have to be disappointed, as is usually the case in this publication. Flat, correctly damped, undistorted bass down to well below 20 Hz is just that; it can only sound one way. Once you've heard it you know it—and you never again want to be without it. It brings you the real world of music, not a preshrunk facsimile. And now, for the first time, it costs relatively little. If you have a listening room of reasonable size, nothing can improve your stereo system as dramatically for $750 as the Hsu Research HRSW10. •

"Qui ne gueule pas la vérité, quand il sait la vérité, se fait le complice des menteurs et des faussaires."

He who does not bellow the truth when he knows the truth makes himself the accomplice of liars and forgers.

—CHARLES PÉGUY (1873-1914)

In Your Ear


IT HAS FLAT FREQUENCY RESPONSE FROM THIRTY TO FIFTEEN THOUSAND CYCLES!
class of 1953
THAT'S PING-PONG STEREO, MAN! I STILL LIKE MONO.
CLASS OF 1963
THE HIGHS ARE GRAINY AND THERE'S NO FRONT-TO-BACK DEPTH.
CLASS OF 1973
DIGITAL SUCKS. A GOOD ANALOG LP BLOWS IT AWAY.
CLASS OF 1983
CLASS OF 1993
AUDIO? WHO CARES? I'M GETTING WINDOWS 3.1 FOR MY PC.

Nostalgia and Loudspeakers

By Drew Daniels

My response to a hundred requests for the lost sound of yesteryear.

Editor's Note: *Drew Daniels was until recently Principal Electroacoustic Engineer at Walt Disney Imagineering. Before then he was the Applications Engineer for Pro Sound products at JBL Professional. He is one of the leaders of the Los Angeles section of the Audio Engineering Society; he is the author of various AES technical papers; in other words, he is a genuine pro, not a retired dentist who likes to build speakers in his hobby shop.*

* * *

Memory.

Nostalgia isn't what it used to be. Our recollection is proactive. My 1970 Webster's doesn't include that word, but the newer one says it's a psychological term coined in 1933. It's an adjective meaning "relating to, caused by, or being interference between previous learning and the recall or performance of later learning ([as in] proactive inhibition of memory)."

Memory is not a constant but, rather, a fuzzy collection of impressions swimming in the chemical reactions of the brain. There are strong, precise recollection processes like recalling facts while watching *Jeopardy,* and then there are indistinct tenuous "recollections" of things like odors, touch sensations, colors and sounds. But, to be specific, even the fuzzier recollections of sound can be divided into strong and weak recollections. For example, the sounds of familiar voices on the phone form strong enough recollections to allow us to recognize people quickly, while the timbre of individual voices is virtually impossible to recall except by a small number of people who are "tuned-in" to timbre, such as impressionists, who use it in their work .

A test for the presence of B.S.

If you know any pompous "experts" who claim to possess a calibrated ear, there is an easy way to humble these golden ears. Send five bucks to Riverbank Labs in Geneva, Illinois, and get yourself one of their fine milled aircraft-aluminum tuning forks. I have found that the A-880 works well for this test. Strike the fork and place it an inch from one ear, then quickly move it to the other ear and back and forth. The object is this: People are living organisms (most people), not test equipment. Most of the time a person's two ears don't hear the same way. At some times, the left ear will hear a pitch as flat or sharp compared to the right ear at that same time. At other times, this pitch perception is reversed, or the two ears hear the same pitch, depending on brain activity and the blood pressure in the individual ears, which can vary depending on mood, head orientation, or even facial expression. If you fail to obtain a different pitch perception with this little test, wait a minute or two and then repeat the test. Most likely, the result will be different perceived pitches.

Anyone who claims to be "calibrated" or have machine-like abilities of hearing is either lying or mentally incompetent. We are not machines, nor can we be.

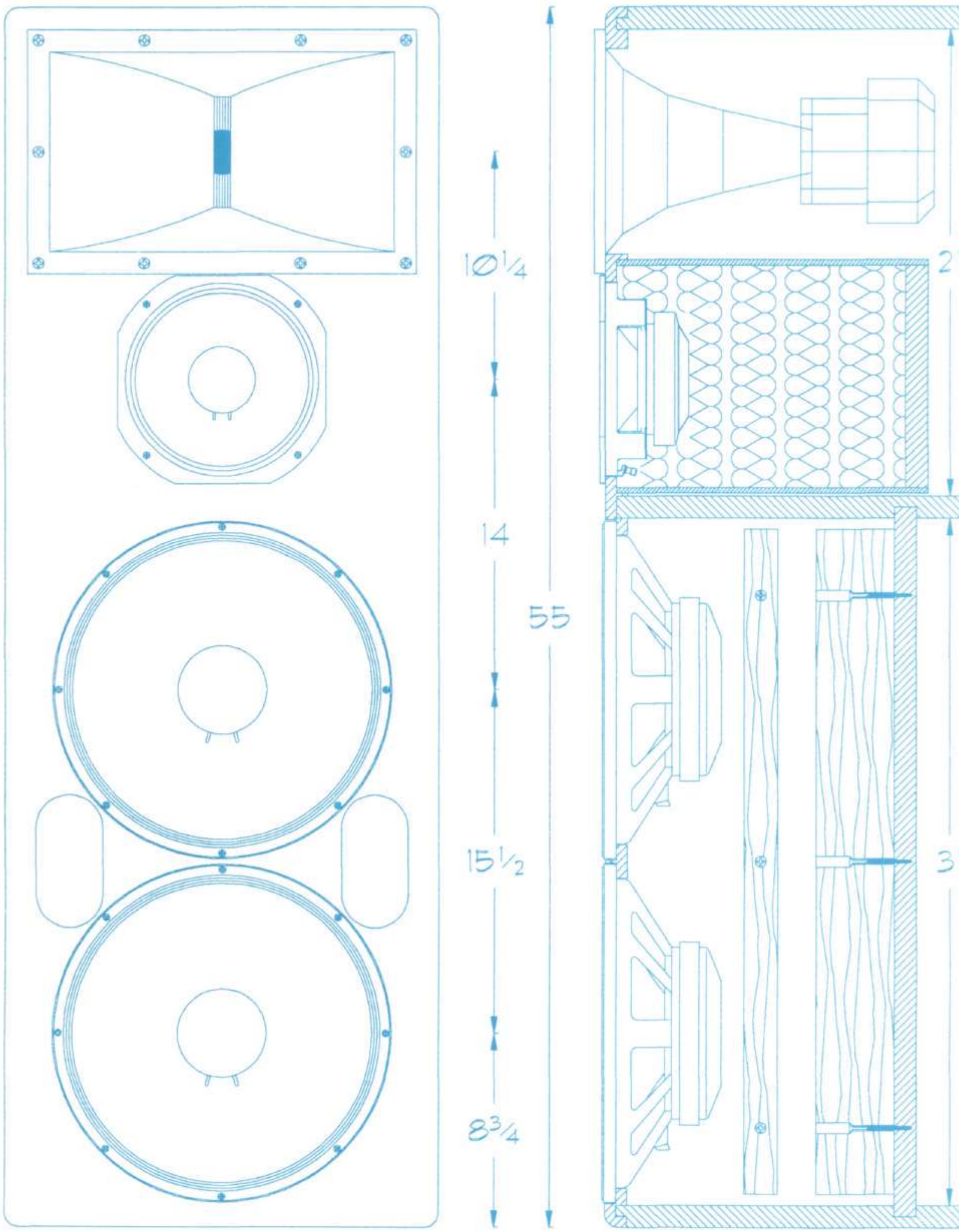
The request.

During the five years I was JBL's Applications Engineer, many audio "old-timers" called looking for information about old speaker systems. After just a few months on the job, I began to realize that their interest in these old systems was often a considered and educated personal listening choice, established with good knowledge of more modern "advances" in loudspeaker technology. So strong and adamant was their insistence that the old systems were simply better and more musical than any of today's systems—including the modern $40,000+ supersystems—that I began to experiment with some of the components, and with some of the older systems when I came across them.

As a result of five years of such experiments, computer modeling, culling and matching of components and, of course, listening, I'm convinced that the old systems and their technology contained the germ of the correct approach to music reproduction using loudspeakers. Modern devices are available to add the pieces needed to complete what they started—provided you are not trendy, have plenty of cash lying around, and have an understanding family and neighbors.

Some explanatory background.

Audio power amplifiers these days can be very economical indeed. If the purchaser isn't too fussy about designer labels, a power amp can be had for as low as a dollar per watt—even less in the larger economy sizes—and this represents, at least in my mind, one of the best values one can obtain in the audio marketplace. As a mat-



Basic configuration and dimensions of the main speaker (without subwoofer) discussed in the article.

ter of engineering policy, I generally recommend that the largest amp available, affordable or practical, be used in a given system. This is because the largest amplifier design within a manufacturer's line is usually the flagship model and, more importantly, usually has more output devices (transistors) and greater capability to deliver large amounts of current into low-impedance loads.

One of the amplifier designer's most difficult challenges is second-guessing what kind of load his product might face once it ends up in the hands of the end user. This is a fact that would seem to argue for speakers with purpose-built amplifiers attached, but since most audio consumers tend to think they can get better sound by selecting their own components, the idea has met with marketing failure.

In general, an amplifier selected for low output resistance, high damping, low distortion, and so on, will be fine for most speaker loads it may encounter. This amplifier type will produce the least variation in system response due to the slings and arrows that amplifiers are subject to.

When I was a kid and my ears were at their best and my musical sophistication near nil (ca. 1955-57), I was easily impressed by sounds that were simply unusual, that is, the sound of a system with "crisp" highs, since this was something one didn't find very often in the mainstream of audio systems. Low bass from loudspeakers was utterly unknown—subwoofers were not yet even a concept. About the best one could ever expect to hear were systems way beyond the economic reach of consumers, namely theater systems with enormous enclosures, large recording studio monitors, and the like. There were a few "home" models made from the available pro-style components in furniture-quality cabinetry that made them semitolerable to understanding, or deferential, '50s wives.

Of course, in the late '50s amplifiers available to the nonprofessional public were small in size, mostly for price reasons, but even "professional" amplifiers designed for industrial installations only provided between 5 and 30 watts, with the exception of the new wave of "super amplifiers" boasting a gigantic 60 watts. Speaker manufacturers were making voice coil bobbins out of ordinary kraft paper, and the adhesives used to hold speaker moving assemblies together were unsophisticated, not yet having benefited from the great advances science and engineering were to enjoy in the coming "Space Age."

This period, in loudspeaker history, is very important for a number of reasons. First, decisions were made that would guide the form designs would take and affect all subsequent progress right up to the present. These decisions took speaker design in the obvious direction of taking advantage of the falling prices of amplifier power, thanks to the introduction and implementation of the transistor. With plentiful available power, designers could virtually ignore speaker efficiency. They could add mass to speaker moving assemblies to drive down resonant frequencies and reduce driver size, allowing them to stuff pretty good bandwidth into small boxes. A case in point might be the early AR series bookshelf systems from Acoustic Research, which offered efficiency well below 1%—as opposed to the larger counterparts that could offer up to 20% efficiency figures. Further, as adhesives improved, loudspeaker drivers became more rugged, inviting amplifier manufacturers to keep increasing power, and this technological spiral continues still.

Second, the "High Fidelity" movement/industry really began in earnest, as compared to the esoteric hobbyist status it had occupied through its early history.

Third, the technologies I referred to earlier made possible things that would not have been considered before, opening up the loudspeaker design field to methods that would eventually provide inexpensive products of astounding performance-per-dollar value.

Today, we are basically stuck with such high-value speakers. I say we are "stuck" with this type of speaker, but let's understand why this is so. It has become easy and inexpensive (read that as high profit margin) to make one-inch dome tweeters that perform very nicely, thank you. It has become easy to mill speaker enclosures automatically from pallet-loads of lumber—allowing the boxes to be shaped to take advantage of the shape of the lumber supply to get maximum yield and minimum waste. It has become easy to stuff into these mass-produced boxes loudspeaker components specifically designed to operate near optimally in the box volumes provided, thanks to computer optimization for the needs of the model, its size, sensitivity, bandwidth and price. The prices are held low by reducing or shortening worker operations, as you would expect. I have seen some small, moderately priced speakers being made in a total of about 200 seconds!

Looking fondly at the past.

I used to be an "audiophile." Thankfully, I've outgrown it. When I realized I wasn't enjoying music anymore—just playing with hardware—I mostly abandoned the pursuit of system perfection, put together a passable system, and began listening again. My ears, however, still search for a sense of realism, for which I really have only one criterion: the output (from the speakers) should sound like the input (the live acoustic instruments). The system should be transparent to the source material and particularly to the performance.

I was educated in music. I studied opera in the *bel canto* voice style, performed roles in college, light opera, played in big bands and folk groups in the '60s, and rock and jazz groups through the present. If there's one thing I can say from a musician's point of view, it's that music from loudspeakers is not right! More to the point, music from loudspeakers can never be right. The sound that we hear is the result of all the limitations speaker designers feel constrained to put up with. Despite all the progress in materials and techniques over the years that people have devoted to making speakers, the results are all trade-offs.

They have to be, because musical instruments don't have dispersion characteristics like speakers. Pop music recordings are mostly monaural anyway. Everything is panned up the middle with some stereo reverb, so it really doesn't much matter what a mass-market speaker does as far as "imaging" is concerned, since there's nothing to image in most recorded material. In the case of pop music, it's worth remembering that the sound placed on the recording is the art form itself. The sound of the individual instruments is often obtained by shoving microphones into the instrument to get signal-to-noise ratio improvement, decrease mike-to-mike leakage, or to produce a microphone-proximity-effect bass boost, or to reduce ambient sound from the room. All this forms a part of the pop recordist's art but ends up preventing listeners from hearing anything like an image of the instruments that produced the recorded sounds. As far as actual "stereo" is concerned, there is only volume level panning if you're lucky, and only some artificial electronic reverberation if the recordings are typical pop. (See my article on this subject in the November/December 1991 issue of *db Magazine.)*

Having played in an orchestra and a big band, I can tell you firsthand that what comes out of speaker-reproduced recordings of these ensembles bears little resemblance to what happened originally.

The fidelity of reproduction to the original sound source has always been the heart and soul and the core reason engineers pursue audio fidelity, notwithstanding Sony MiniDiscs, Philips Digital Compact Cassettes and other forms of squashed audio—but that's another tirade for another article. *[We'll see how "squashed" they are when we test them.—Ed.]*

And now, the nitty-gritty.

During the course of progress toward producing high-quality, low-cost loudspeaker systems for consumers, designers have ignored what I consider to be the worst and most important form of distortion, *dynamic distortion.* Dynamic distortion in loudspeakers is caused by the instantaneous local heating of the voice coil wire. This is not the global motor-structure heating cited as the cause of power compression in large, professional drivers, but the very local heating that takes place in the smallest volume of wire, that volume which is comparable to the size of the wire cross section. If you were able to obtain an infrared detector sufficiently sensitive at long IR wavelengths and point it at a voice coil, you would find that the IR (heat) output of the wire would be somewhat proportional to the audio power input to the wire. I say somewhat, because there is thermal inertia, and the transfer function is not perfect. This imperfect transfer function is also nonlinear, and gets worse as higher power levels are applied to the wire and global heating takes place. The essence of this phenomenon is that power pumped into a loudspeaker is lost in nonuniform proportions to the input power. *[This explanation is new to me and my associates. It may be perfectly valid, but I would have preferred to see it confirmed here by an authoritative reference or two.—Ed.]*

How to improve this situation? Well, one way is to use less power. Make less heat. But if we use less power, we will need more efficient transducers.

Now, in the world of touring sound companies, theaters, discos, stadiums and other big sound installations, designers follow the seemingly human principle that if you pay a lot for a speaker, it should be able to deliver endlessly higher volume without burning up or breaking. We know this is a flawed principle, but the attitude persists. And so, pro loudspeaker manufacturers keep trying to supply drivers that won't break under standard operating conditions, namely abuse. I suppose, as an academic or engineering exercise, it's valuable to make drivers of this sort; after all, if it can handle a sufficient amount of power, then it can support added mass and lower resonance, and be used as the low end of a two-way system, making it cost-effective and relatively small—both reasonable and marketable attributes.

Without constraints.

If we now decide that the idea of a highly efficient loudspeaker system is interesting and that the sound could prove satisfying, we must look at the trade-offs (remember, *there are always trade-offs).*

The first trade-off to deal with is driver bandwidth. Higher efficiency, less bandwidth. It's like putting a turbocharger on your car's engine; you get more torque, but over a narrower range of engine revolution speed. If we choose high-efficiency drivers, they will individually cover narrow frequency bands, requiring perhaps 4-way or 5-way systems, instead of 2-way or 3-way.

The second trade-off is that to achieve high efficiency, driver motor structures need the highest possible magnetic flux, which means enormous and/or expensive magnetic assemblies; also, driver motors must not waste any resource—no extra mass such as overhanging voice coil windings. This means that coils and air gaps will be close to the same size, allowing for little or no linear coil excursion. On the positive side, excursion in this situation causes even-order distortion, which sounds musical, adds brightness, and in speech systems that are driven hard, produces an increase in speech intelligibility. The third and fourth trade-offs, as alluded to before, are expense and size.

High-efficiency systems will need larger drivers and will press air volume into service. Remember, there is a good reason why bass fiddles and tubas are bigger than violins and trumpets.

There is also a rather nontrivial drawback to building a high-efficiency speaker system that is only fair to tell you about. There is no such thing as a high-efficiency subwoofer driver. You obtain high efficiency in the low bass range by coupling lots of cones together and moving lots of air. You will want a subwoofer to match the rest of the system, and I'm warning you right now, before you get ready to allocate space and funds, that you're probably looking at four 18" woofers in about 50 cubic feet of box, and a 1200-watt amplifier to drive them, in order to keep up with the system I'm about to describe. You may never play it loud enough to justify four drivers, but I include them as a recommendation, because my goal in building this system was to keep distortion below 1% at any musically realistic playback level, at any frequency in the human hearing range.

I have chosen my system drivers to be close to the spirit of the old-technology drivers. JBL has kept this spirit alive in some of their models, the 2123, 2202, 2220, the "E" series "MI" speakers, and of course their compression drivers. The drivers are:

(4) 2245H subwoofers (2.1% conversion efficiency each or 8.4% in tandem)

(4) 2220J high-efficiency 15" woofers (one pair each side, about 17% efficient each pair)

(2) 2123H high-efficiency 10" midranges (3.5% efficient each)

(2) 2382A horns with 2450J compression drivers (30% efficient each)

By efficiency, I mean for example, that if you put 100 electrical watts in, you will get 30 acoustical watts out, as in the case of the horn.

If you're wondering, can the poor 10" mid with a mere 3.5% efficiency keep up with the horn, rest assured, I needed 10 dB of attenuation on the mid to get flat frequency response.

For amplification, I used two BGW SPA-3 triamplifiers built for me by BGW to provide highpass filtering for the two fifteens, lowpass and horn EQ filtering, switched attenuation, and built-in delay to acoustically align the cones and compression driver. It turned out to be an elegant and simple alternative to a large rack of gear. Although the amplifiers each take up only 5¼ inches of rack space, they each produce up to a total output of 1000 watts, providing 600 watts for each pair of fifteens, 200 watts for the mid and 100 watts for the horn (because the compression driver is 16 ohms). This represents an average of over 30 dB of headroom above normal living-room listening levels, which generally range in *milliwatts* for these speakers.

To give you a better idea of the difference in efficiency I'm describing here—between low-efficiency audiophile-type speakers and these high-efficiency speakers—for example, if you played, say, Magnepans at 100 mW, you would have a hard time hearing and understanding speech from your listening seat, particularly if your refrigerator is running in the kitchen or if your kids are on the phone. At a one-watt average input, the Magnepans will be producing what you would most likely regard as background music. At the same one watt into the high-efficiency speakers, you will find some musical passages to be uncomfortably loud, and your spouse will be moved to exclaim, "lower that fi!"

Theory of operation.

I will labor each point here because there are likely to be some statements I make that will require justification in the mind of those readers who have never been exposed to the type of technology or components I'm describing.

Horns.

First, let me tell you why and how I believe horns have acquired their unjustly poor reputation. The first reason is their cost; the second, their size; the third, their traditionally intended application; and the fourth, their misapplication. It has been some years now since speaker system manufacturers have offered horns in systems intended for sale to home users. Real horns, not schlock-shack plastic piezo types, are designed to impart directional characteristics to the output of compression-driver transducers. Such transducers are acoustic transformers, and generally represent science and experimentation of the highest order when they are designed and executed correctly. It has taken over 40 years, aerospace-style dynamic finite-element analysis, the highest-tech materials, ten-year development cycles (I'm talking careers here), and endless tweaking and testing to get the near-theoretical performance the largest "professional" compression drivers provide. Proper horns couple the low acoustic impedance output of the compression driver to the air we sit and listen in. If the designs are good, the coupling provides output which is the same as the input, that is, the frequency response in the so-called acoustic free-field is the same as the frequency response of the compression driver when it is mounted on a terminated tube (totally absorbing transmission line). In addition, and in concert with the goal of achieving this technically non-trivial ideal of operation, horns can only control the dispersion pattern of sound when the horn dimensions are larger than the sound wavelengths. As the mouth or the depth of the horn gets smaller with respect to the increasing wavelength of progressively lower frequencies, the dispersion begins to "bloom out" and tend toward omnidirectional. A rule of thumb I use for a lot of my "off the top of the head" engineering is that the dispersion will shrink to about 90 degrees (at the -6 dB off-axis angles) when the wavelength is the same size as the source transducer it's coming from. You can see right away that if, for example, you want to control the horn's dispersion down to 1000 Hz (wavelength 1.13 feet), the horn should be over a foot in mouth size. This makes for big, often unacceptably industrial-looking devices occupying the wall space on either side of the TV. This is a perception not ignored by most speaker manufacturers' marketing departments.

In my opinion, many if not most speaker systems are marketed on the concept of good sound from small-sized packages. This necessarily causes the design of such products to rely on drivers which are small and have small voice coils, which, as I mentioned earlier, heat up too much and produce that worst form of distortion, dynamic distortion.

To use a horn properly in an extremely high-fidelity speaker system, it must be transparent to the sounds it will be asked to reproduce. This seems an obvious statement, but it begs explanation and dispelling of the reputation for "horn sound" or "honkiness" so common among all but the cognoscenti. The explanation is that old cliche, "trust me" (I try to avoid cliches like the plague), but a thorough explanation could easily fill a succinctly written 2200-page college text book and is still the subject of scholarly investigation by engineers and physicists doing their doctoral thesis.

The essence of the explanation is this: Horns that have a directional control over the sound they are producing exhibit what engineers call a "Q factor," which is a number used to describe how the output angle differs from omnidirectional. *[Don't confuse this with the Q of a resonant circuit!—Ed.]* For example, if we place an omnidirectional loudspeaker in free space, at the top of a 10-story flagpole, it would radiate everywhere with the same radiation intensity (Q=l). If we place the same omnidirectional loudspeaker on a hard reflecting surface, it's radiation will be forced to reflect into half the space it did on top of the pole (Q=2). To an observer or a measurement microphone, the loudspeaker will behave as though it has twice the acoustic power output, or 3 dB more than it did on the pole.

Horns force their output radiation into fairly narrow angles, such as 90 degrees horizontally and 40 degrees vertically, and can thus achieve acoustic gain in trade for more omnidirectional sound dispersion. If the Q factor of a horn is 10, then it will produce about 10 dB more acoustic output on axis (where it's aimed) than would an omnidirectional device with the same original acoustic power output. This allows us to get extremely high sound levels with very little power relative to, say, a dome tweeter.

There are some tricks that are essential for eliminating horn "honk." The first is to use the cone driver placed just below the horn all the way up to a frequency where it begins to "beam" in accordance with the relationship of sound wavelength and cone diameter. At a frequency where the resulting Q factor of the cone matches that of the horn, the transition from cone to horn will be smooth, and not abrupt—as it can be in systems where the cone is too large and the horn is too small. If this condition is met, and the frequency response of the cone is good well beyond the frequency up to which it is used (a well-behaved upper-end rolloff), then the horn will enjoy a seamless transition from the cone and will not honk, assuming its frequency response is good and uniform over its output angle. This latter condition is referred to as being "power-flat" and is very important to the transparent operation of the speaker system in rooms, with their concomitant acoustic implications. If the speaker system is power-flat, the sound in the room will be as good as that particular room will allow it to be.

The midrange.

The midrange driver must be a cone, unless you live in a theater and don't mind a 5-foot high horn (I crossed my mid at 300 Hz into the woofers). As it turns out, a mid cone supplying 300 Hz to 1200 Hz gives the proper effortlessness with very little power, and thus has extremely small cone excursions and low distortion. As I mentioned earlier, I had to trim the 2123H mid cone back 10 dB on the amp's gain control to get the response through the band flat. The power absorbed by the mid cone driver amounts to milliwatts most of the time, which helps to hold harmonic distortion to very low levels.

The driver is mounted on the baffle as close to the horn as I could get it with my inexpensive mid-chamber geometry. You could do better if you are willing to cut out the lower lip of the horn and snug the mid frame up into the cutout, and figure out a mid-chamber arrangement that would clear the horn and driver behind the baffle, but this is not measurably better than just a touching fit. The enclosure for the mid cone consists of a 10-inch diameter concrete casting tube made of plasticized paper. The tube is mounted to the baffle by gluing into a counter-bored shoulder cutout, routed in the back of the baffle around the mounting hole. The tube is about 12 inches long (deep); it is filled completely but loosely with a "jelly roll" of fiberglass cut from a roll about 4 feet long. The back end of the tube is sealed with a disk of 1-inch medium-density fiberboard—the same material used to build the rest of the box.

I experimented with a dozen midrange drivers before I was confident that the 2123H with its high efficiency and limited excursion linearity would produce sufficiently low distortion. It is a wonderfully transparent driver and a large part of the reason why this speaker system sounds like listening to live music rather than loudspeakers.

The woofers.

The 2220 fifteen-inch cone driver should be thought of as a low-midrange, not really a woofer. Yes, it's a big driver with a big voice coil. In fact, I use two of them in my bass guitar rig, but its Q_{TS} and moving mass are so low that when you put it through Thiele-Small calculations and plot curves on a computer, as I did a hundred times, the device ends up looking more like a midrange itself. To be accurate, the Keele exponent-corrected program I use (because it tends to give me systems that measure the same as the model predictions) calls for about 1.5 cubic feet per driver, tuned to about 85 Hz—not exactly organ pedals. I ended up opting for a slight overdamping of two units in a 3-cubic-foot volume tuned to 80 Hz. Even so, the unassisted output of the box is flat to 65 Hz and droops only slightly at 40 Hz, in the middle of a 40,000-cubic-foot room. *[I didn't know you lived in Windsor Castle, Drew.—Ed.]*

You must make sure the highpass filter on the amplifier is set to 80 Hz and rolls off at a rate of at least 18 dB per octave. The 2220J drivers are high-efficiency, limited linear excursion devices (in fact, they are one of the highest efficiency cone drivers made anywhere). The crossover frequency of 80 Hz is the design target to limit cone excursion and produce a good transition to the subwoofers.

The enclosure is built to be as rigid and nonresonant as possible and then lined with fiberglass over the entire interior surface area, except around the ports, where air turbulence might spray fiberglass around. The woofer portion of the enclosure is the only real structure. The midrange tube is extremely rigid and exceptionally nonresonant. My goal in designing the woofer section was to minimize spurious panel vibration and acoustic output—within reason. There is more panel output, in fact, from the back cover of the compression driver and horn walls.

I used four two-by-fours for bracing inside the woofer compartment. These were counterdrilled for wood screws and glued on-edge to the compartment panel interior surfaces. I tried to space the braces at random—so that no two unbraced panel areas were the same size. I also glued the two cutout disks from the woofer holes to the back panels to make the total panel thickness 2 inches, plus braces!

Setting up the system.

This can be tricky. I used a TEF analyzer and a $1/_{24}$-octave Brüel & Kjær real-time analyzer. First I did energy-time measurements to set the delays in the amplifier. This proved to be difficult, since I had to take distance-ranging measurements of each driver separately to make sure I was looking at arrival times from the intended measurement object. After I got the delays set, I checked frequency response the best I could in the space available, then resorted to the real-time analyzer. Be aware that the frequency responses you get with these two methods are very different because TEF windows its measurements to try to exclude reflections and examine only direct sound from the source, while real-time analysis includes all returning room energy information along with that from the speaker. I like a balance of both measurement methods, because one lets you fine-tune the energy output of the speaker, and the other lets you adjust large trends like the general "too-bright" high end you will likely notice if you try to obtain flat output to 20 kHz from the direct-sound readings of a truly power-flat or "constant-directivity" horn. Such horns can produce more high-frequency energy in the first place, and then they can deliver it to even fairly large rooms. Most people are not used to listening to a power-flat top end and will find it too brassy. Only minimal-miked big-band recordings will sound right with the system adjusted that way.

For your playback system, I recommend that you use an extremely low-noise preamp with simple Baxandall-type "tone controls." The bass and treble turnover frequencies are a matter of taste, but 100 Hz and 10 kHz seem to work well to adjust this system to music on recordings. To my ear, parametric equalization is less pleasing and is certainly prone to putting phase aberrations in more audible frequency ranges than are simple tone controls. One-third-octave "graphic" equalizers are completely useless for high-fidelity use and one-octave units are even worse. *[How about a Cello Audio Palette? If you can afford the speaker, you can afford the Cello.—Ed.]*

You should start adjusting with the amp gains set low and turn the preamp all the way up. Slowly advance the amp gains until you can achieve proper balance and slightly louder than necessary output. This will ensure that you will have the least possible amplifier hiss from the speakers. Amplifier hiss is a phenomenon that rarely troubles owners of low-efficiency speakers, but these monsters are efficient enough to make the transistor junction noise of poorly designed amplifiers quite audible.

If any of you reading this decide to build this system, it will cost around $10,000, including amps and all. If you get that serious, if you are rich and adventurous and don't care what stereo salesmen think, you are the type of person who would really enjoy this system. *[Rich? I know poor audiophiles who have accumulated five-figure stereo systems. But maybe that's why they're poor.—Ed]*

The subwoofers should generally occupy the space between your main speaker systems. Because of the ear's forgiveness, you'll find there's a "window" of space for physical placement that allows a good deal of flexibility in fitting the speakers into your listening space.

What you have when you're done.

These loudspeaker systems could easily be used behind a perforated movie screen, providing sound to an audience of hundreds of people in a small movie theater. They are also equally capable of causing you, in a home listening-room setting, permanent hearing loss, and doing so quickly.

I have two tips for you, and you would do well to pay heed:

First, play music at no more than realistic levels. I assume that if you choose to build these things, you've done so because you're interested in the fidelity of the reproduced sound to the originally recorded sound. You will get the best representation of the original sound if you play the reproduction at the original sound level. Playing too loud is as detrimental to fidelity as playing too softly.

If you play predominantly rock music, you need to keep a sound level meter handy. You can get a perfectly adequate one at your local Radio Shack store for around thirty bucks, and for your ears' sake, don't ignore this advice. These speakers make so little distortion that you will be tempted to believe that the 120 dB sound you are listening to is only playing at 90 dB. *This is not good. You will lose your hearing. Don't let this happen!*

If you find that the clean sound causes your favorite

(continued on page 53)

Miscellaneous Electronics for Audio and Video

By Peter Aczel
Editor and Publisher
&
David A. Rich, Ph.D.
Contributing Technical Editor

Not really an article but a catchall assortment of individual reviews, by two editors, of unrelated or loosely related audio components, grouped together here for mere convenience.

As our regular readers surely understand by now, and our new readers soon will, we at *The Audio Critic* expect no audible differences in our tests between high-quality preamplifiers, amplifiers, and other linear analog electronics, and report such differences, if and when discovered, with some degree of surprise. That's one of the main differences between us and the golden-eared high-end journals, where they just know that the $6000 unit will sound better than the $1500 one and, of course, confirm that expectation in their tests. We believe, on the other hand, that an audible difference always has a *cause* and that in the absence of an explicable cause any report of an audible difference is highly suspect—unless verified by a series of double-blind comparisons at matched levels (and who does that except us and archvillains like David Clark and Ken Pohlmann?).

Thus the emphasis in the applicable reviews that follow is on measurable accuracy and quality of construction. Accuracy—i.e., transparency to the signal—so we won't even have to think about the electronic component when evaluating the sound of the total system that includes it; quality of construction, so that the accuracy can be expected to continue for a long, long time. We talk specifically about the sound of the electronic component only if there's something new or different to be said. All pretty obvious.

—Ed.

Full-Function Preamplifier

Bryston 11B

(Reviewed by Peter Aczel)

Bryston Ltd., 57 Westmore Drive, Rexdale, Ont., Canada M9V 3Y6. Model 11B preamplifier, $1450.00 with optional balanced outputs. Tested sample on loan from manufacturer.

The difference between this model and the lower-priced Bryston .5B reviewed in Issue No. 18 is greater flexibility and additional features; the basic engineering concept, circuit design, and build quality are exactly the same and require no further reviewing. In essence, this is merely an editorial follow-up, without new insights by David Rich.

Model 11B is actually the same preamp as Model 12B, which David Rich mentioned in passing as a possible recommendation at the end of his series of reviews; the difference is only that the 12B incorporates a moving-coil step-up transformer, whereas the 11B I tested does not. Any 11B can be retrofitted, however, to become a 12B. Together these models represent Chris Russell's current thinking on top-of-the-line preamps—and in my book he is the most levelheaded thinker in this crazy mixed-up industry.

The 11B has more inputs and outputs than the .5B; it can accommodate two tape recorders, although the tape outputs are still unbuffered, so the same caveats apply regarding a connected tape deck which is not powered up. (Bryston has recently added specific instructions about this.) In addition, the 11B I tested had both unbalanced and balanced outputs. The substantial price difference you must pay for all this is explained by the superb construction of the expanded switching facilities and additional connectors. (Remember, Bryston products are guaranteed for 20 years.)

The basic Audio Precision measurements on the 11B yielded results virtually identical to those reported in the .5B review and need not be repeated here. Two exceptions: (1) Channel separation in the line stage was greatly improved, though still not very impressive; I measured -44 dB cross talk at 20 kHz in the less good channel, dropping at 6 dB per octave to -102 dB at 20 Hz. That's a 12 dB improvement across the board, the result of recent work on the printed-circuit board. (2) The balanced output adds a tiny bit of distortion because of the additional active stage; the minimum THD plus noise of the line amplifier was about 6 dB higher through the balanced output than through the unbalanced (main) output. The message is clear: use the balanced output only when the power amplifier is far away and the interconnect is so long that the signal needs extra protection—or when the power amp has only a balanced input. (Sorry, tweaks, I couldn't hear the difference. Can you hear the difference between -88 dB and -94 dB distortion? Yeah, right.) It should be added that the balanced output clips at 30 V, as against 15 V for the unbalanced output.

In actual use this preamp is truly a joy—no hum, no hiss, no pops, no funny noises of any kind, just smooth operation and flawless sound. That total endorsement applies only to the current version; an earlier production sample gave me some minor problems that are no longer relevant. I think even the most finicky audiophile would be satisfied with the 11B, except perhaps for its price, which is quite high but not high enough to impress some insecure high-end gurus. My regular standby, the Boulder MS, has perhaps even better measurements by just a hair (in some but not all categories) and a few nonessential extra features (polarity inversion, mono/stereo switching, channel reversal, and such) plus modular construction—but it costs so much more! If I didn't already have a world-class preamp, I know I could live happily with the Bryston 11B or 12B in my main system.

Cable Enhancer (it says here)

Duo-Tech Model CE-1000

(Reviewed by Peter Aczel)

Duo-Tech Corp., 37396 Ruben Lane, Building F, Sandy, OR 97055. Model CE-1000 Cable Enhancer, $179.00. Tested sample on loan from manufacturer.

Tweako cult items in audio fall into four broad categories: (1) ridiculously priced but gets the job done—e.g., silver cable; (2) ridiculously priced and screws up the job—e.g., UltrAmp D/A Converter; (3) ridiculously priced and accomplishes nothing—e.g., Tice clock; and (4) fairly priced and accomplishes nothing—e.g., this so-called cable enhancer. When you take it apart and look inside it, you see electronic circuitry and parts that should sell for approximately as much as they ask for the product. That's good. When you examine the manufacturer's claims and rationale for it and run some simple tests, you realize that it's pure B.S. That's bad. I don't think there's fraudulent intent behind the Duo-Tech, just utter silliness and self-deception.

I'm sure that many of our readers have already heard of this magic device. The tweako testimonials and anecdotal raves have been out there for a while. (Hey, there's never a shortage of wishful thinkers in the audio world.) In case the whole thing is new to you, here's the gist of it: You get this cute little chassis, about as big as a ten-pack of 3.5-inch computer diskettes. It has a pair of phono jacks at each end and a front panel with two switches and two LEDs. You also get two plug-in adapter modules that convert the phono jacks into speaker-cable terminals, plus an AC power adaptor that supplies the 12 volts DC used by the unit. To "enhance" a pair of cables, you connect them between the jacks/terminals on the left and right, energize the system, and let the cables "burn in" for at least 48 hours, or more for "quality cables" (it says in the manual). The "signal flow direction" is terribly important; if you have plebeian cables without directional arrows on them, the treatment permanently locks in their directionality (from left jacks/terminals to right), and you must label them with little arrow stickers that come in the package. (How's that for anal, Sigmund?) The following before/after differences in the sound are claimed (are you ready?): Better focus, improved coherence and dynamics, reduced glare and ring, more realistic soundstage, better defined and tighter bass, more extended highs. What about "burning in" with music instead? Not nearly as good!

I was of course curious what kind of signal has the extraordinary capability of producing such results, so the first thing I did was to connect an oscilloscope across the signal path of the device. Well, it's a pulse train consisting of squarish pulses of completely random duration but equal amplitude, about 14 V from peak to peak, the tops and bottoms steeply tilted as if by highpass filtering. I could see no difference when I switched from the Interconnect mode to the Loudspeaker mode. I must add that I lost very little sleep trying to pry further into the secrets of the design; it's some kind of simple digital circuit.

Then I went out and bought two brand-new pairs of identical 3-foot interconnects. Nothing fancy but of decent quality, with good shielding and nice gold-plated plugs, the kind a noncultist like me is happy with. I took one pair and burned it in (if you'll pardon the expression once again) for 96 hours on the Duo-Tech. I left the other pair untouched. Then I did my listening test. I plugged both pairs into my trusty Boulder MS preamplifier, so that I could switch from one to the other with a single click of the source selector switch. I plugged the other end of one pair into the line output of the Sony CDP-X779ES CD player (the best in my tests) and left the remaining plugs of the other pair dangling free near the line output, ready to be swapped instantly. Thus I was able to switch back and forth between the burned-in and the virgin pair of cables in about two seconds while a CD was playing—not a blind test, to be sure, but I was ready to set up a formal ABX comparison in case I *thought* I heard the slightest difference. But—you guessed it!—I heard no difference whatsoever on any kind of music, even though I knew which cable was which, so how could I possibly have heard a difference under double-blind conditions? And how could anyone, under any condition, have heard a difference that had no reason within the laws of physics to exist? Afterwards, I felt a little sheepish for having spent time and energy on such a silly exercise—but somebody had to do it, right?

I flatly refuse to dignify the muddleheaded techno-babble offered by Duo-Tech as the scientific rationale for the Cable Enhancer by printing it and then refuting it here. Instead, I'll make the same offer as I did to George Tice regarding his magic clock. If Duo-Tech can produce three electronics experts with university graduate degrees in engineering or physics who are not commercially linked to the company or its products and who will certify in writing that Duo-Tech's claims for the Cable Enhancer are scientifically valid, then I shall devote a special issue of *The Audio Critic* exclusively to the explanation and celebration of the Cable Enhancer technology and mail it as a free bonus to all subscribers. Any bets on the outcome of my offer?

52" Rear-Projection TV with Surround Sound

Magnavox RM8564A

(Reviewed by Peter Aczel)

Philips Consumer Electronics Company, One Philips Drive, P.O. Box 14810, Knoxville, TN 37914-1810. Home Video Theatre 1992: Magnavox RM8564A, with JBL speaker system and Dolby Pro Logic Surround Sound, $3400.00. Tested sample on loan from manufacturer.

Of the three available large-size TV formats—direct view, rear projection, front projection—I strongly lean toward rear projection at this time. Even the 35" direct-view CRTs are too small for maximum viewer involvement in, say, football or opera (to name only two of the many kinds of panoramic TV fare), whereas the unquestionable impact of the big front-projection format is canceled out by the immense inconvenience it creates in any room that isn't exclusively dedicated to home theater viewing. A 52" rear-projection set is big enough to satisfy my craving for lifelike dimensions in audiovisual entertainment, yet it still fits into a multipurpose family room in which kids, dogs, friends, neighbors, etc., come and go. Admittedly, the resolution of detail isn't quite as high as on, say, a 19" Trinitron, but the overall presentation imitates life more convincingly because of the scale. (See Issue No. 12 on the subject of small versus large video.)

This particular Magnavox unit is not only an excellent rear-projection TV but also unusually sophisticated in terms of audio, interesting enough to be reviewed in an audio journal. Unfortunately, by the time this is in print, it will be on its way out of the stores, to be replaced by a 1993 Philips model (Philips and Magnavox being virtually interchangeable brands, like Dodge and Plymouth at Chrysler). Unlike high-end audio components, TVs in all price brackets undergo a yearly model change, leaving a slowpoke reviewer like me panting as I bring up the rear. No matter; it's the current Philips approach to home theater packaging that's under review here, not just a 1992 model.

Rear projection, not long ago rather low in video fidelity, has been getting a great deal better lately, and the Magnavox RM8564A illustrates just how much better. The picture isn't *significantly* less good than on a 32" or 35" direct-view set, and that's quite something. (No, I didn't say it was *as* good.) The trade-off is between the highest possible definition and the greatest possible realism in dimensional perception. As I said, I like the trade-off. Let me, however, discuss the audio system first, as it is the long suit of the set. It would actually pass as an upper-mid-fi home music system by itself, without the video, and that's more than I can say for any other TV known to me.

On each side of the 52" screen, there is—get this!—a JBL pure-titanium 1" dome tweeter, the same I enthused about at some length in Issue No. 14. I never expected to find it in a TV, I must say. Below the screen, on each side, a JBL long-throw 8" woofer in a vented enclosure is crossed over passively to the tweeter to form a very creditable full-range speaker system, with response from about 40 Hz to well beyond 20 kHz. A 25-watt amplifier channel drives each side; in addition, there are two more 25-watt amplifier channels available for two surround-sound speakers in the rear.

There is a wide range of stereo and surround-sound modes to choose from. The simplest is plain stereo out of the two built-in speaker systems. The most elaborate is full Dolby Pro Logic, which requires an additional stereo amplifier feeding two speakers flanking the TV set plus two rear speakers fed from the built-in extra channels. The set's own speakers and their amplifiers then become the Dolby center channel. Other combinations between these two extremes can be easily set up, depending on the available program material and ancillary equipment. In each case, the audio quality is just as good as you would expect it to be with a component system using similar amplifier power and the same surround speakers. Home theater can—and should—sound just as good as component audio of comparable sophistication.

The microprocessor-controlled audio and video settings that can be selected from the screen menus of this set are as varied, elaborate, and versatile as any I have seen and then some—not really surprising when you consider Philips's deep involvement in microprocessor technology. Remote control jockeys will surely experience an unprecedented sense of power as they issue all those push-button commands. One small complaint: to switch between program sources (antenna, cable, VCR, laser disc, etc.) requires going to a submenu of a submenu of the main menu. That's a bit ridiculous; program source selection is one of the most frequent user operations, and most remote controls have separate buttons for it. Some totally unrealistic computer nerd must have thought up that one—and I don't even think it has been fixed in the 1993 successor model. Once you get used to it, though, it matters very little.

To get back to video quality—that's why you buy an expensive TV, after all—the first thing you notice about this rear-projection system is that you can't view it too far off to the side or standing up. The set's really excellent picture quality is evident only when you sit in front of it or at a small angle to it. I would say that four front-row seats are the maximum for proper viewing. Of course, you can always set up a second row. Again, no big deal, but viewers accustomed exclusively to direct-view CRTs are initially bewildered.

As usual, I put the set through its video paces with the various excellent tests on the Reference Recordings laser videodisc *A Video Standard.* Black level retention (i.e., the ability to hold black at black, regardless of the picture content) was quite good for consumer equipment but not studio-perfect—and I didn't expect it to be. Contrast is not the long suit of rear-projection TVs; the peak linear capability of just about every set will be exceeded before the contrast is as high as one would ideally like it to be, and such was the case here. The default setting by the factory was a very acceptable compromise, however. Color performance via the S-video input was very good, and the factory settings of Color and Tint were pretty much on the money. Geometry was also quite precise, with no significant distortion on checkerboard patterns, circles, etc. Convergence as adjusted at the factory needed no trimming. On a subjective basis, the picture with a good program source was truly beautiful in color and had excellent definition in my opinion—always keeping in mind that it's rear projection on a large screen.

The advertised horizontal resolution of the set is 600+ lines, but that's a nebulous area of specifications where *laissez-faire* reigns—with my best laser disc player and the RR test disc I can resolve only about 400 lines, and the Magnavox easily passes that test. (The theoretical best for NTSC broadcasts is 336 lines.) More interesting is the set's PIP (picture-in-picture) capability: you can, for example, watch your videocassette of *Terminator 2* (or do you prefer *Wings of Desire?)* and at the same time tune in the news on CNN within a cutout in the main picture. Unfortunately, you can't simultaneously view two programs from set's own tuner; you need an outside source for the second program.

The 1993 successor model to the Maganavox RM8564A is the Philips 52NP51FS. The price is the same; the amplifier wattage has been increased and the audio functions expanded (among other things, you no longer need an outboard amplifier for full Dolby Pro Logic, although you can add one for even more power); the video circuitry and optics are also claimed to be improved (800+ lines of horizontal resolution is the new spec!?). It looks like an even better deal to me, except that the new 1" tweeter and 8" woofer are by Philips, not JBL, so I can't predict the speaker performance.

To sum up, if you like large-screen rear-projection TV as much as I do—and that's a purely personal preference—then give serious consideration to the rear-projection home-theater packages by Philips (under its various brand names) because they offer the rare combination of excellent video *and* audio performance, with options for integration into even more elaborate home audio systems.

Line-Level Preamplifier

Monarchy Audio Model 10

(Reviewed by David Rich)

Monarchy International, Inc., 380 Swift Avenue, Unit 21, South San Francisco, CA 94080. Model 10 Buffered Control Center, $980.00. Tested sample on loan from manufacturer.

This preamp is an example of the best of science and the worst of tweako loony tunes. The most significant difference between this preamp and pure tweako products is its relatively moderate price ($980).

Mechanical construction is excellent. The front panel is massive, the sheet metal thick, and the metal volume control knob is 4 cm in diameter and 2 cm deep. The high-quality RCA jacks are directly mounted to the rear panel. The PC boards are double-sided with plated-through holes. The RCA jacks are wired directly to selector switches, which are directly wired to the volume control. The amount of hand wiring in this unit significantly adds to the construction cost. The wire is claimed to be pure silver. I consider silver wire an extravagant waste of money, but it could not have added very much to the cost of this preamp given the unit's reasonable price. The volume control is a huge (4 cm in diameter and 9 cm deep) stepped attenuator manufactured by Shallco. The control, which would be more at home in a spacecraft than a preamp, has 30 steps, each 2 dB. This is the most expensive control I have encountered in a commercial preamp, including those priced in the high four figures. A balance control is not included on this preamplifier. The switches are of the finest quality I have seen on a preamp, being both sealed and gold-plated.

The selector switch arrangement is a dim-bulb tweako cultist idea. A pair of inputs is directly connected to a three-way toggle switch. The up position puts one of the two input signals on the bus, the down position the other; the middle position disconnects the switch from the bus. A complement of three toggle switches is used for the six inputs of this preamp. It is possible in this arrangement to connect more than one input signal to the bus. Once, when we engaged the preamp in this fault condition, the line amplifier went into a latch state, rendering the preamp inoperative until the power supply plug was pulled (the unit has no power switch), but we were subsequently unable to recreate this. The preamp will consistently go into a latch mode if the power supply is interrupted briefly. This is the result of the power supplies collapsing asymmetrically. The problem will be fixed in production units according to the manufacturer (we tested a very early sample). Believe it or not, this $980 preamp has no tape monitor function—tweak-tweak!

The line stage uses an Analog Devices AD744 BiFET IC for voltage gain, in conjunction with an 8-transistor discrete unity-gain output stage. The AD744 is a high-performance device with a slewing threshold of 0.9 V and a 13 MHz gain-bandwidth product. The op-amp is used in a novel topology developed by Walt Jung, which bypasses the AD744's output stage. A DC servo is used to reduce the preamp's output offset. The PC board on our preproduction test sample had a significant amount of component rework on it. The manufacturer claims this will be corrected in production units. Parts quality on the board is typical for a unit in this price class. In true tweako cultist fashion, a muting relay to prevent power-up pulses from passing to the power amplifier is not included.

The power supply is dual mono, including the transformers. An AC line filter is included in this unit, so you do not have to spend megabucks on an audiophile-approved external one. A separate button quad rectifier is used for the positive and negative supplies of each channel. The unregulated filter caps are 4700 µF, and the regulated filter caps are 8200 µF. LM317 and LM337 IC regulators are used. The regulators have an output voltage of +24 V. 78L18 and 79L18 regulators are used to subregulate the power supplies to the ICs down to ±18V. A pair of subregulators is used for each channel.

Performance of the preamp was for the most part exemplary. Channel separation changed at 6 dB per octave from 110 dB at 20 Hz to 50 dB at 20 kHz. Clipping occurred at 12 V rms. Distortion reached a minimum of -94 dB at 6 V rms across the entire audible band. Switching to a 600-ohm load did not change these results. The state-of-the-art performance of the line amp should drive adherents of the tweako camp away in droves, since tweaks equate good numbers with bad sound.

Clearly this preamp cannot be recommended unconditionally for the reasons outlined above, despite the fact that it represents exceptional value. If a power switch and muting relay were added, and the loony-tune input selector switches were replaced with a rotary function switch and rotary record selector switch, we would have a product that could be unconditionally recommended.

Full-Function Preamplifier

Rotel RC-980BX

(Reviewed by David Rich)

Rotel of America, P.O. Box 653, Buffalo, NY 14240. RC-980BX preamplifier, $499.90. Tested sample on loan from distributor.

This preamp is at a lower price point than the preamps we looked at in the last issue. It is made in Japan but it is very different from most Japanese designs, which typically have a more complex control panel than a 747. The Rotel has just four plastic controls on the front panel—power switch, volume control, source selector (labeled Listening), and record selector (labeled Recording). A headphone jack is also on the front panel. The headphone amp is a JRC4556 op-amp. There is no balance control because at this price point only a low-cost unit could have been selected. The volume control is a 2-cm wide unsealed Alps unit similar to the control used in the much more expensive Adcom GFP-565. The left- and right-channel levels can be adjusted individually. A friction plate on the volume control allows both channels to be adjusted together. The record selector has a novel Off position which disconnects Tape 1 Out and Tape 2 Out from the main bus. This is a poor man's replacement for a tape monitor buffer circuit. Tape 1 Out is disconnected when Tape 1 is selected on the record selector, preventing system damaging oscillations. For an unknown reason there is no Tape 2 position on the record selector, and it is thus impossible to record on tape deck 1 from tape deck 2. The function and record selectors are unsealed linear switches. The front panel has a rotational-to-linear converter which drives the switches through a long unsealed band of metal. This arrangement could be a long-term reliability problem if the band of metal starts to bind or stick in its unsealed channel. The switches are located near the input connectors to minimize jumpers. This is required because the PC board is single-sided. The PC board does not have plated-through holes. The sheet metalwork is slightly thicker than is typical of mass-produced consumer electronics. The unit is held together by sheet metal screws.

The power supply is massive for this design. A 8.5-cm diameter, 4-cm high toroid transformer drives a high-current composite bridge rectifier. The unregulated rails are filtered with a 4700 µF capacitor. Power supplies are regulated by series-connected pass transistors which have no global feedback. The bases of the pass transistors are connected to a zener diode reference. Each channel has a separate regulator pass transistor, but both channels share the same diode reference. Heat sinks are not used on the regulator pass transistors. The regulated power supply rail voltage of ±17.4 V is too close to the absolute maximum voltage rating (18 V) for the ICs used in this preamp to insure maximum reliability. Separate 78M15 and 79M15 regulators are used for the headphone amplifier.

The line stage uses an Analog Devices AD711 BiFET op-amp (4 MHz unity gain bandwidth and 0.8 V slewing threshold). The input (C_1) and output (C_3) are 10 µF electrolytic coupling capacitors. The C_2 capacitor is not present in the feedback loop. Time-delayed relays connected to the outputs prevent turn-on transients from reaching the power amp. In our Audio Precision tests, the line stage clipped at 12 V rms. Low-frequency distortion plus noise of the line stage (we always report the "worse" channel) reached -95 dB before clipping, an excellent result. The 20 kHz distortion of the line stage reached a minimum of -83 dB at 2 V rms, rising to -80 dB at clipping. Adding a 600-ohm load to the line stage reduced the clipping level to 5 V rms. The 20 kHz distortion reached a minimum of -73 dB at 0.3 V rms and remained at that level until clipping. The fact that the line stage is based on the AD711 IC and not a more advanced IC or a discrete transistor circuit can be seen from the last three measurements. Channel separation of the line stage is greater than 98 dB below 200 Hz. It then rises 6 dB per octave to 66 dB at 20 kHz. The simplicity of the signal path and the elimination of a balance control are partially responsible for this excellent result.

The phono equalizer is a two-stage design. The first stage is a linear-gain amplifier with gain selection by a switch on the rear panel. The 2120 Hz pole is implemented passively between the first and second stage. The second stage implements the 50 Hz pole and 500 Hz zero of the RIAA equalization curve actively. The first stage uses a discrete differential pair in conjunction with a Texas Instruments TL071CP. The required noise performance of the phono stage could not be achieved with an IC op-amp alone. The TL071 is a low-cost, general-purpose, JFET-input op-amp. The AD711, used in the line stage, is an enhanced version of the same. The TL071's low (3 MHz) unity gain bandwidth restricts its use in high-performance gain stages (it is widely used in mass-market equipment), but it is occasionally used, with acceptable results, as a low-cost unity-gain buffer in high-end designs. As will be seen below, the TL071 performs in an exemplary manner in the function it is assigned in this preamp. A Signetics NE5534AN is used in the second stage. The Signetics part has better noise performance than the AD711 used in the line stage, but it has a lower slewing threshold, making it less desirable to some designers for use in a line stage.

The performance of the phono stage proved to be state-of-the-art. Phono equalization is held to better than ±0.05 dB in both the MM and MC modes. The channels were balanced within 0.05 dB. These are the best results we have ever measured and should embarrass the designers of more costly preamps. The phono stage clipped at 12 V rms, just like the line stage. Low-frequency distortion plus noise was -74 dB and -90 dB for the MC and MM modes, respectively. These numbers are the result of noise, not distortion, since the lowest measured numbers occurred just before the onset of clipping. At 20 kHz the phono stage distortion reached a minimum of -86 dB at 5.5 V rms.

The Rotel RC-980BX lacks the look, feel, and build quality of the best preamps reviewed in Issue No. 18 but it costs significantly less. It is a clear choice over the more expensive Adcom GFP-565 preamp reviewed in that issue. The phono-stage performance is, as I said, right up to the state of the art. The preamp is therefore highly recommended. The only change I would make would be to use a less overbuilt power supply and apply the cost savings to a better-quality IC in the line stage. •

Another Look at Outboard D/A Converters

By David A. Rich, Ph.D.
Contributing Technical Editor
(with an interruption by Peter Aczel, Ed.)

Everybody is doing something different in this product category, but a faultless design at a realistic price still remains to be seen.

This is a continuing chapter in the search for a CD decoder box that can truly produce analog signals at 16-bit resolution. As will be seen below, it is not the final chapter in that search. Readers who find my analysis of these devices a little too technical are directed to my tutorial paper, which appeared in the now out-of-print Issue No. 15. An updated reprint of the article should be available soon.

Please note that the measurements we report below do not check for limit cycle oscillations, which can be a problem with the delta-sigma (1-bit) DACs. The promised examination of limit cycle oscillations in delta-sigma DACs has been postponed to the next issue.

Recently *Stereophile* has begun measuring the jitter of the internal word clock driving the DAC. We have not done this here for three reasons: (1) Jitter on the word clock will cause discrete tones or an increased noise floor at the output of the decoder. Both of these effects will be seen in the THD + N measurements. (2) Power supply noise and digital input signal feedthrough at the DAC may contribute additional jitter from the DACs internal logic, which will be completely missed by this test. (3) The probe placed on the word clock pin will present an additional load to the word clock driver. Since the word clock driver is not designed to drive the additional load, a false reading can result.

Audio Alchemy DTI/XDP/PS2

Audio Alchemy, 30879 Thousand Oaks Boulevard, Suite 222, Westlake Village, CA 91362. Digital Transmission Interface (DTI), $349.00. Extended Digital Processor (XDP), $300.00. Power Station Two (PS2), $120.00. Tested samples on loan from manufacturer.

These three separate components add up to one D/A decoder box. As will be seen below, Audio Alchemy uses this à la carte approach to increase the potential number of users of its products. All the extra boxes and cables required add to the cost of producing the Audio Alchemy D/A system. One annoying problem is that the feet of the XDP are not high enough for the XDP's front panel to sit flush with the DTFs front panel.

The Power Station Two consists of separate analog and digital power supplies. Two power transformers, two DIP-sized full-wave rectifiers, and four 4700 µF capacitors create the unregulated supply rails. The digital rails are then regulated with 7808 and 7908 regulators and further filtered with 4700 µF capacitors. Heat sinks are used on these regulators and all regulators in the DTI and XDP. Analog regulation is identical, except that 7818 and 7918 regulators are used. The DTI uses the digital supply only. The XDP uses both supplies. The Power Station Two can also be used as an upgraded power supply for some other Audio Alchemy products .

The DTI (Digital Transmission Interface) incorporates an S/PDIF decoder, an S/PDIF encoder, a switch and circuitry to invert the phase of the digital signal, and coaxial as well as optical inputs. The S/PDIF decoder is the Crystal Semiconductor CS8412-CP. This is the same chip as EAD uses in their DSP-7000 processor. While this chip provides good jitter performance, Crystal Semiconductor data sheets show that it can be improved with the addition of a VCXO (voltage-controlled crystal oscillator) based PLL (phase-locked loop) placed after this chip. Audio Alchemy does not include this second PLL. A very early press release for this unit indicated it would use a GaAs (gallium arsenide) based VCO running at a very high clock rate. The VCO clock was then to have been divided down to the system clock frequency. The division process reduces jitter from the VCO. This intriguing approach is, unfortunately, not used in the production version of the DTI.

The DTI has two outputs: (1) a modified I^2S bus output and (2) an S/PDIF output. The I^2S bus is one of a series of data transmission bus standards used to connect chips in a CD player or D/A decoder. I^2S has no advantage over other bus standards. The I^2S bus does not transmit subcode information; thus Audio Alchemy adds an additional signal line to the I^2S bus to transmit the de-emphasis code. The S/PDIF output is generated from the I^{2}S bus data by a Crystal Semiconductor CS8402 S/PDIF transmission chip. Separate 7805 subregulators are used for the main digital circuitry, the PLL, and the data receiver and transmitter.

The process of receiving the S/PDIF data and then retransmitting it is claimed to reduce jitter levels when the signal is ultimately decoded in another D/A decoder box. While this approach has the potential to reduce the jitter levels of the auxiliary decoder box, it cannot eliminate jitter completely. To understand what the DTI can and cannot do we need to introduce some basic concepts of digital data transmission. Given the limited space I have to do this here, and to prevent the MEGO (My Eyes Glaze Over) effect, we will use very simplistic models.

For a digital data stream to be meaningful, it must be accompanied by a clock signal which partitions each data bit cell. Thus a minimum of two wires is required, one which contains the clock and one which contains the data. To represent the clock and the data in a single wire, an encoding method must be used. The method used in the S/PDIF format is called biphase mark. In this encoding scheme, a clock transition always occurs at the beginning of a bit cell. If a 7 is to be transmitted, a clock transition occurs in the middle of the bit cell as well. Note that even if a long string of 0s or 7s is transmitted, at least one clock transition will occur during each system clock period. These added clock transitions allow the S/PDIF decoder to recover the system clock from the single wire. One method to decode the reference clock is to use a PLL. A PLL can be thought of as a flywheel. The encoded digital signal sets the speed of the flywheel. The speed of the flywheel's rotation represents the PLL's output—the recovered clock. One rotation represents one clock cycle. The flywheel will continue to operate at the clock frequency even if the clock transitions are not always present. Now we have three sources of error: (1) Short-term changes in the speed of the flywheel can occur when the clock transitions are absent. Recall that the number of clock transitions per cell is data-dependent. (2) A jittery incoming data stream can cause short-term changes in the flywheel's speed. (3) The flywheel's speed may change randomly as a result of deviations from ideal operation. (Flutter would be an example in our mechanical analogy; a noisy VCO in the actual PLL.) Now, error sources (1) and (3) occur in the PLL of the S/PDIF decoder independently of the jitter in the incoming data stream. Only error source (2) can be reduced by reducing the jitter in that data stream. Error sources (1) and (3) can only be reduced by improving the performance of the S/PDIF decoder.

Please note that the encoded S/PDIF signal is not a pure digital signal. The performance of the S/PDIF decoder can be degraded if the signal is distorted by the data transmitter, through the S/PDIF cable, or at the data receiver. It is thus possible to see performance degradation in some S/PDIF decoder designs if the communication channel is not optimum. Inexpensive solutions are available to make sure that the S/PDIF signal is not distorted. Very expensive glass fiber or coaxial cables are not required for optimal performance. Better S/PDIF decoders are less sensitive to waveform distortions in the received S/PDIF signal. As I have explained previously, the best solution is to eliminate the problems associated with the single-wire encoded signal and use a two-wire system. This approach is used in the newly introduced Denon DA-X D/A converter. Audio Alchemy had the option to do this too, since they also make a CD transport, but they chose not to do so.

The XDP (Extended Digital Processor) converts the I^{2}S bus data into an analog signal. An earlier Audio Alchemy product, the DDE, also generates the I^{2}S bus data required by the XDP. First the I^{2}S data is passed to a Burr-Brown DF1700P digital filter. (This is a remarked version of the NPC SM5813. See Issue No. 15 for more details on the NPC part. NPC has just introduced a second-generation filter chip, the SM5842AP. More details on this chip will appear in the next issue.) The output of the digital filter goes to the Philips SAA7350 delta modulator D/A chip (again, see Issue No. 15 for more details on this chip). This chip generates a one-bit data stream, which is filtered by the Philips TDA1547 chip (see Issue No. 16, page 45, for more details on this chip). The output of the TDA1547 is balanced. It is converted to a single-ended signal by a discrete six-transistor operational amplifier stage. Finally, the signal passes through a second-order Sallen-Key filter which uses a two-transistor unity-gain buffer. The buffer is an *npn* emitter follower biased by a current stage formed with a *pnp* transistor. Separate supply subregulators are used for the digital chips (a 78M05) and the TDA1547 (two 79M05s, one for the analog and one for the digital section of the chip, and a 78M05.), and separate regulators for the left and right channels of the discrete output stage (78M15 and 79M15). This brings the total number of regulators to 15! All this circuitry just fits on the double-sided PC board. The board is clearly not the product of a garage operation, since the parts are autoinserted and the board has plated-through holes (the DTI board has similar construction). The de-emphasis is performed by a relay. Power-up transients are passed directly to the output, since no muting circuit is included. Small noises would also occur at the output of the unit under some conditions when the S/PDIF signal was interrupted. LEDs on the front panel indicate lock and de-emphasis. It is also possible to put the decoder in the de-emphasis mode with a front panel switch. The DDE device does not have the extra de-emphasis code at its I^{2}S bus output; thus the de-emphasis must be engaged manually when the DDE is used.

Measured performance of the DTI/XDP combo was a for the most part very good, but a major problem was identified in the distortion tests. The frequency response was down by 0.3 dB at 20 kHz. For a disc with de-emphasis, the frequency response is down 0.5 dB at 20 kHz. Channel separation was 120 dB or better below 200 Hz. It then decreased at a 6 dB per octave rate to 80 dB at 20 kHz. With a bigger chassis a better result might have been possible at 20 kHz. Noise spectrum analysis showed no hum components down to -125 dB. This shows that the external power supplies and the extensive regulation are achieving the desired result. Gain linearity remained within ±0.2 dB down to -100 dB. Full-scale THD + N proved to be disappointing; from 20 Hz to 1 kHz it was -87 dB and it then rose to a maximum of -77 dB at 10 kHz before dropping again as the harmonics began to fall into the stopband of the reconstruction filter. Reducing the signal level by 20 dB resulted in a reduction of the THD + N to -95 dB relative to a full-scale signal. (Referenced to the fundamental signal level this figure differs, of course, by 20 dB.) At the reduced signal level the analog output stage is no longer contributing distortion components to the output signal. Noise from the analog electronics and nonideal performance of the DAC itself account for the remaining 3 dB deviation from the theoretical minimum of -98.08 dB. The poor full-scale performance is very likely the fault of the two-transistor buffer circuit. Perhaps a more complex buffer could not be fitted into the small enclosure.

The distortion of the buffer circuit prevents an unconditional recommendation of these Audio Alchemy units. The performance was more than adequate to insure that the unit has no audible colorations, and ABX test results confirmed this. If all the Audio Alchemy units were combined on one 19-inch chassis, then the total cost of the unit could be reduced and the buffer stage could be improved. Such a unit would receive a high recommendation by this journal.

EAD DSP-7000

(follow-up)

(By the original reviewer, Peter Aczel)

Enlightened Audio Designs Corp., 607 West Broadway, Fairfield, IA 52556. DSP-7000 outboard D/A converter, $1399.00. Tested sample on loan from manufacturer.

The main reason for this follow-up is the muddle-headed, irresponsible review of the EAD DSP-7000 by Robert Harley in the September 1992 issue of *Stereophile* (Vol. 15, No. 9). Believe it or not, the man reports 20 dB *more* noise across the audio spectrum with a digital zero input than with a dithered 1 kHz input at -90.31 dB. That's like saying that the rainfall was 2 inches more under the roof of the carport than in the driveway. On top of it, he actually notes that this *is* peculiar and then—standing there with his bare face hanging out—he comments, "Whether this correlates with the disappointing sound quality remains to be seen." Reading that rubbish I had little doubt that what it correlates with is Harley's questionable qualifications as an audio equipment reviewer, but just to prove my point I thought, what the hell, I'll ask the nice people at EAD to send me another sample of the DSP-7000. This time I got a factory-sealed box out of production stock, containing a unit with the more restrained black front panel I had wished for.

Well, what do you know, when I measured the new unit on the Audio Precision "System One Dual Domain," the noise spectrum from 30 Hz to 200 kHz with digital zero input ranged, in the less good channel, from -140 dB on the bottom end to -117 dB at 20 kHz and -93 dB near the limit on top. That's 34 dB better on the bottom, 27 dB better at 20 kHz, and at least 13 dB better in the ultrasonic region than Harley's less good channel. Let's face it, Bob, old buddy, you messed up bigtime. There's nothing wrong with the noise floor of the DSP-7000.

While I was at it, I investigated the THD + N versus frequency performance of this unit in somewhat greater depth than in my original evaluation (see Issue No. 17). It turns out that the digital circuitry is capable of literally perfect 16-bit resolution but the analog circuitry introduces a teensy bit of nonlinearity at maximum output. The evidence is that with an input of -20 dB, the THD + N as normalized to 0 dB (full scale) reads almost exactly the theoretical minimum of -98.08 dB across the entire audio spectrum, up to where the analog filter begins to roll off the output. With a 0 dB input, however, the reading ranges from -95 dB at 20 Hz to -90 dB between 3 kHz and 11 kHz, and that difference has to be the contribution of the analog circuitry. These results are slightly better than what I obtained with the first sample; the newer production unit appears to be free from the 60 Hz problem I reported at the time. *(Stereophile,* by the way, never reports THD + N versus frequency in digital playback equipment. No test is more important, but the ultrahigh-end stuff doesn't always pass it with flying colors, so it becomes unmentionable.)

Gain linearity in my new sample was off by +0.5 dB at -80 dB and +1.5 dB at -90 dB; however, as David Rich explained long ago, gain linearity specs must not be confused with the more important and stringent integral and differential linearity criteria. Channel separation ranged from 130 dB at 20 Hz to 82 dB at 20 kHz, changing at 6 dB per octave. (Again, Harley's measurements were considerably worse.) A minor flaw I overlooked in my original review was the not quite dead-accurate de-emphasis, with -0.2 dB error at 10 kHz and -0.4 dB at 16 kHz. That, however, includes the inherent top-end rolloff of the unit, which is almost certainly due to the analog filter and amounts to -0.07 dB at 10 kHz and -0.2 dB at 16 kHz. I could not hear any "softening" of the highs as a result; it takes a larger fraction of one dB up there to be reliably audible.

Thus the EAD DSP-7000 can be declared to be close to perfect in the digital domain, only ever so slightly imperfect in the analog domain, and unusually full-featured as well as nice to look at in the user domain. It matters very little to me at this point exactly how much of a "breakthrough" the so-called AccuLinear *I-to-V* converter is. The radiated RFI problem, on the other hand, is still as bad as ever; my FM tuner, sitting four feet from the DSP-7000, chirped like a bird when the latter was either on standby or fully turned on, but especially when on standby. The remedy was to pull the plug.

The next version of the EAD DSP-7000 will use the 20-bit Burr-Brown PCM63P-K DAC instead of the 20-bit Analog Devices chip in the original design. An upgrade kit is also being made available. I expect the low-level gain linearity measurement to be essentially perfect with the new chip, among other benefits.

One other interesting little sidelight. EAD's research director, John Hagelin, the Harvard Ph.D. who originally conceived the design of this D/A processor, was the presidential candidate of the Natural Law Party in the recent national election. You may have seen the TV commercials. The Natural Law Party advocates, among a number of other things, transcendental meditation as a solution to the problems of this messed-up world. And that's not all. Dr. Hagelin currently teaches at The Maharishi International University in Iowa, where transcendental meditation is the unifying discipline under the leadership of the Maharishi Mahesh Yogi. Alastair Roxburgh, the Englishman who is engineering director of EAD, is also associated with the university. Now, this is an audio magazine and not a forum for the discussion of political or spiritual matters, but isn't this fascinating? Here is a far-out group with a pretty extreme belief system—and I say that merely as an outside observer, without any judgmental implications whatsoever—and yet they are doing solid engineering work based on mainstream science without tweako distractions. How strange, then, that there are Episcopalian Republicans out there who believe in CD rings and silver cable!

Monarchy Audio Model 22A

Monarchy International, Inc., 380 Swift Avenue, Unit 21, South San Francisco, CA 94080. Model 22A Dual 20-Bit D/A Converter, $1200.00. Tested sample on loan from manufacturer.

This unit represents one of the first affordable D/A processors to use the Burr-Brown PCM63P-K multibit DAC. Of all available monolithic DACs, this chip has the lowest specified distortion levels as published in its data sheet. A novel topology (see Issue No. 16, page 49) allows the DAC to have excellent linearity around digital zero. It is also used in the $4000 Theta DS Pro Generation III. Monarchy hired a team of highly qualified consultants to aid in the design of this unit. Different consultants were involved in the design of the S/PDIF decoder, the DACs' peripheral circuitry, and the analog section. The PC board is double-sided and the board has plated-through holes. The components on the board are autoinserted, and modern surface mounting is used for the S/PDIF decoder. Extensive attempts to limit RFI radiation, including an AC line filter, are part of the design of the 22A. Sheet metal work is of a high quality. A phase inversion switch is on the front panel, but there is no lock or de-emphasis indicator. In addition, the unit has no power switch. The unit accepts both optical and coaxial inputs, switchable from the front panel.

The power supply consists of separate analog and digital power supplies. The digital transformer has two secondaries. Separate button-sized full-wave rectifiers and 15,000 µF capacitors are used to for the positive and negative digital supplies. A 78M05 regulator is used to supply the digital circuits. Separate 78M05 and 7905 regulators drive the PCM63P. The analog supply consists of another button regulator and two 8200 µF capacitors on the unregulated supply rails. An RC1515 dual-tracking regulator is used for the ±15 V supplies. I am not familiar with this part but it appears similar to the Raytheon RC4195. A separate 78M05 regulator connected to the analog transformer is used to power the analog supply of the S/PDIF decoder. All regulated rails have either 330 µF or 100 µF capcacitors connected to ground.

The S/PDIF decoder is a new second-generation chip from Yamaha, the YM3436C. The first-generation Yamaha chip, the YM3623B, is often maligned by Robert Harley in *Stereophile.* Contrary to the assertions by Mr. Harley, the YM3623B can provide good performance if properly applied (the Yamaha CX-1000 preamplifier is a good example). The preliminary data sheet for the YM3436 does not give specifications for its recovered clock jitter. The output of the YM3436C is passed to a Burr-Brown DF1700P digital filter. The current-to-voltage converter is the new Analog Devices AD811 high-speed transimpedance amplifier (2500 V/µs slew rate, 60 dB PSRR at 100 kHz, 65 ns settling time to 0.01%, and 140 MHz bandwidth). To get this speed, the chip burns a half watt, and Monarchy Audio uses a heat sink on the chip. Two PMI OP275 op-amps (9 MHz bandwidth, 7 nV/ Hz voltage noise at 30 Hz, and a slewing threshold of 0.4 V) are used in the analog filter stage. As explained in Issue No. 18 (page 37), this op-amp uses a composite bipolar/JFET input stage, which attempts to combine the low noise of a nondegenerated bipolar input stage with the high slewing threshold of a JFET op-amp. While the OP275 is a good op-amp, I would like to see a state-of-the-art op-amp such as the AD797 used instead in the 22A, given its high price point. The OP275 operational amplifiers are used to form a GlC-based (generalized impedance converter) reconstruction filter. A Burr-Brown application note has shown that this filter topology has the potential for slightly lower noise levels in comparison with the Salien-Key topology, but it is nearly twice as complex. The gain of the filter stage can be changed by 6 dB by changing shorting links on the PC board. An open-loop discrete buffer follows the filter. I prefer the closed-loop arrangement used in the same company's Model 10 preamp to this open-loop approach. A DC servo prevents direct current from appearing at the output of this unit. Another OP275 is used to drive the balanced outputs. A transistor switch connects the passive de-emphasis network. No muting relay is used at the output, a significant omission in my opinion. Power supply transients were small, however, and surprisingly no transients occurred when the S/PDIF signal was connected or removed in a variety of nefarious ways.

The measured performance of the Model 22A was a mixed bag. The frequency response was tipped up by 0.1 dB at 20 kHz. For a disc requiring de-emphasis, the maximum frequency response error was +0.2 dB. Channel separation was 110 dB or better below 3 kHz, then decreased to 98 dB at 20 kHz. Noise spectrum analysis showed significant power-supply components at multiples of 60 Hz; the 300 Hz component was particularly strong at 96 dB below full scale.. In addition, the overall noise floor appeared higher than required to achieve 16-bit signal-to-noise ratios. Gain linearity remained within ±0.4 dB down to -100 dB. Full-scale THD + N proved to be disappointing. From 20 Hz to 20 kHz, it remained flat at -85 dB. Lowering the input signal level by 20 dB or even 40 dB does not change this result. This indicates that noise, not the distortion, is causing the degradation in signal-to-noise ratio. All analog components used in the 22A are specified to have noise levels significantly lower than what we measured; thus I cannot explain the origin of the noise.

The disappointingly high noise level prevents a strong recommendation of this unit. Monarchy is investigating the cause of the noise and will hopefully be able to lower it to acceptable levels, since the rest of the design appears to be excellent. A follow-up review will be forthcoming if a revised unit is made available for test.

UltrAmp D/A Converter

Mobile Fidelity Sound Lab, 105 Morris Street, Sebastopol, CA 95472. UltrAmp D/A Converter, $1295.00. Tested sample on loan from manufacturer.

This is one of the three units in the Michael Yee designer collection (his signature is on the back of each unit). This processor accepts only coaxial inputs. It has a phase inversion switch on the front panel. For some unknown reason, the power switch is located on the rear of the unit but at least this function is included. The PC board is a high-quality double-sided board with plated-through holes. Two inexpensive transformers are used for the analog and digital supplies. Three DIP-sized full-wave rectifiers are on the board. A single 7805 regulator services all the digital circuitry. A separate 79M05 generates the DAC's -5 V supply. All analog circuitry shares the same master-slave topology regulator; 7818 and 7815 devices are used in the positive regulator; 7918 and 7915 devices are used in the negative regulator; 2200 µF filter capacitors are used on the regulated and unregulated rails. A muting relay circuit is included in this design but it does not work correctly. Power-down transients are still passed directly to the output. This is the same problem we also had on Mr. Yee's preamp.

The S/PDIF decoder is the new Philips SAA7274P. Unlike other S/PDIF decoders, the SAA7274P is designed with much of the analog circuitry for it off the chip. This gives the potential for the generation of a very low-jitter word clock. The phase detector and loop filter are fully balanced, increasing the PLL's ability to ignore power supply noise. Some of the advantage is lost in Mr. Yee's design, since a single regulator is used to service all circuits operating at 5 V. The high point of the Philips design is the crystal-based voltage-controlled oscillator (VCXO) that can be implemented with the SAA7274P. The high effective Q of the crystal insures that the VCO will contribute a very small amount of jitter. Mr. Yee does not use the crystal, even though a space was created for it on his PC board. An inductor replaces the crystal in a last-minute board change. The digital filter is the Philips SAA7220P/B, and the DAC is the Philips TDA1541A S1 "Golden Crown." These old parts were last used in state-of-the-art CD players at the end of the Reagan administration. Mr. Yee says they "sound better" than modern chips. The I/V converter and analog filter stages appear to be similar in topology to the poorly performing circuitry used in the UltrAmp preamp. We never did get schematics from UltrAmp.

The measured performance of the UltrAmp processor was worse than what we would have expected to measure in a $200 mass-market CD player. The frequency response starts to decline at 2 kHz and is down by 2.5 dB—yes, 2.5 dB!—at 20 kHz. This is of course quite audible. Channel separation was 100 dB below 1 kHz. Above that frequency it decreased at a rate of 6 dB per octave to 70 dB at 20 kHz. Noise spectrum analysis showed significant hum components. The 60 Hz component was at -88 dB relative to full scale and the 180 Hz component was at -93 dB. Gain linearity error was -1.5 dB at -80 dB and -4 dB at -90dB. This is an out-of-spec result for a "Golden Crown" TDA1541A S1 DAC. Apparently no incoming QC is done by UltrAmp. Full-scale THD + N was -80 dB from 20 Hz to 10 kHz and then rose to a maximum of -67 dB at 20 kHz. This is equivalent to the distortion at 20 kHz that would result if the data were encoded to just 11 bits. Would you want your name to be on the back of this unit if you designed it and it performed this badly?

In the last issue we promised a listening evaluation of the complete three-component UltrAmp hookup, but this was abandoned when the frequency response errors in this unit were discovered. The UltrAmp D/A converter is a very expensive treble control. Clearly, for all of the foregoing reasons, it cannot be recommended. •

From Tweak to Geek
or Satanic Realism

By Tom Nousaine

Editor's Note: *Tom Nousaine, strange visitor from the planet of the Southeast Michigan Woofer and Tweeter Marching Society, came to the audio world with objectivistic powers and abilities far beyond those of mortal audiophiles. Disguised during working hours as a mild-mannered specialist for a great metropolitan telephone company, he fights in his spare time a never ending battle for truth, justice and the scientific way of looking at audio. The plan is to make this column one of our regular features. Our readers have probably seen Tom's name in a number of small audio journals; as a matter of fact, he has told the "Julian story" in print before, but this is the complete, unabridged, philosophically appended version. It's a story every audiophile should be exposed to, no matter where he sees it first.*

* * *

I finally realized at the last CES that I no longer want to be called an audiophile, tweak or purist. Those neat terms, once an endearment to me, have been usurped by people who want us to spend needless thousands on unproven spikes, wires, pillows, clamps, clocks, tubes and other weirdness. The same people violently react when asked to prove their claims.

Let me expand. I will do practically anything to improve my sound system. A quick look at my eight-channel (10 channels of amplification) Lexicon CP-3 system with 23-driver dipole line arrays, 22-cubic-foot tubular 18" subwoofer, and 5-way rear surround speakers will attest to this. But, I still have to take abuse from friends, the underground press and people hawking ridiculous products and ideas, about "refusing to just listen for myself and being "insensitive" or "uncaring about music." I beg your pardon?

When the Evil Gunslinger said "I'm faster than you" to the Sundance Kid, the Kid said "Draw, sucker!" When Bob Falfa said "I'm aimin' to blow his [John Milner's] ass off the road," Milner dropped the clutch. Why then do audio salesmen and manufacturers get away with not having to verify their claims?

Oh, people have "just listened" and heard the news you say? Yeah, well why does the magic disappear when the blindfolds appear? Why can't some audiophiles prove their claims when listener bias controls are employed? When that happens I say it's because the claims aren't real.

Now, many illusions that that go away when the lights come on can be considered to be real. Stereo for example is a perfect example of a "real" illusion. The effect can be duplicated, remains perceptible under controlled conditions and can therefore be verified to others. Anything that can't isn't real enough for me to waste time, money or energy chasing.

And I am getting impatient with those who don't require verification from the proponent or who want to vilify me for asking for it. It just ain't my job to prove the Golden Ear claims. (Even though I have often tried!) If they can't or won't, that's their problem. Would you let a sports car salesman sell you the "fastest car in the valley" if he refused a race? Or claimed that a stopwatch made him too nervous to prove it?

Speaking of proving it, here's a neat little story about a guy named Julian *[not Hirsch—Ed.]* who had the balls to put his beliefs on the line. In the fall of 1991 he told me about how great his new Sumo Andromeda II sounded compared to the Adcom GFA-555 it had replaced. I told him the Sumo was probably a fine piece but I doubted that it sounded much different from the Adcom or any other quality amplifier. Push came to shove and Julian accepted my offer of an ABX double-blind comparison at a Prairie State Audio Construction Society (PSACS) meeting in November 1991.

The Sumo was matched against my $200 (used) Parasound HCA-800II 80-watt-per-channel solid-state amplifier in a "short" system of just a Sony D-15 CD player, the amps, an ABX switchbox and a pair of Dahlquist DQ-lOs. Julian felt the audio character of the Sumo was apparent under open conditions prior to starting. He picked the recordings and controlled the switchpad (which allows an unlimited number of comparisons and unlimited time for each decision or trial) for the first three trials.

After nine decisions he quit the test. Other audiophiles completed the remaining three of the 12 total trials. Results: Julian correctly identified amplifiers 2 out of 9 times (not even once out of the three trials where he picked the records). Best score was 7 out of 12, and the overall correct rate was 48% of about 100 trials. I concluded that when levels were carefully matched in each channel, the Sumo and Parasound sounded exactly alike.

Julian was not impressed. After further discussion he felt the test conditions must have somehow interfered with his ability to tell amplifiers apart. He felt this way even though he had been able to hear the character of the Sumo in the test setup under open conditions and was allowed to tell the other listeners what they should have been hearing. Anyway, Julian got another chance.

On April 18, 1992, I took the Parasound HCA-800II and the ABX switchbox and relay module to Julian's place. I installed the ABX machine and the Parasound in his system, which included a NYAL hybrid preamplifer, JSE Infinite Slope speakers, $700 worth of interconnects and Tara Labs speaker cabling, in addition to the Sumo.

Julian then had as long as he needed to (1) determine how these amplifiers sounded different from each other, (2) determine if the ABX equipment interfered with the revelation of those differences and (3) select music programs which highlighted the differences. By May 16 he had concluded that the amplifiers did sound different from each other and that the ABX equipment did not obscure those differences, and he had selected CDs and LPs that highlighted those differences.

On Saturday, May 16, 1992, I adjusted the level controls on the Parasound to match the Sumo within 0.1 dB at 1 kHz in each channel. I also verified that each of the amplifiers was flat within 0.2 dB from 20 Hz to 20 kHz. No other controls were changed and nothing on the primary system was touched.

We agreed that if he could correctly identify amplifiers 12 out of 16 times using any music he wanted, controlling all the switching and the duration of trials, I would agree that he was right, i.e., the amplifiers were indeed sonically different. We also agreed that the results, no matter what the outcome, would be submitted for publication.

The test began at approximately 1 p.m. The equipment had been turned on the day before and allowed to warm up overnight. I suggested Julian start with the music selectio- where the differences were most apparent. He decided to use the phrase repeat on his CD player for the first selection, so he could use the same music passage for identification. That first trial took about 15 minutes.

Twelve minutes into trial number two, Julian looked up and said, "I have to admit I just can't tell them apart." I asked him for at least 10 trials just to give us enough data to analyze and to assure himself that he didn't leave out any music material where differences indeed could be heard.

Another hour and ten minutes later we had 10 trials. There was no pressure to hurry. Julian had complete control over program selection (including LPs), switching intervals, switching direction and trial duration. And he'd had four weeks of practice with the switchpad.

He correctly identified amplifiers 5 out of 10 times or at the rate expected if he were just guessing. And he was just guessing by his own admission. He didn't get either of the first two right, by the way. Thus, a confirmed longtime audiophile, choosing his own music, controlling every variable in his personal reference system with a month-long warm-up, was unable to correctly identify his $1599 Sumo Andromeda II compared with a $200 (used) transistor-piece-of-crap Parasound HCA-800II when the faceplates were, in effect, removed.

Now there were two small but interesting technical differences between these amplifiers. The Sumo had a 2 dB channel imbalance. Further, the voltage sensitivities of the two amplifiers were markedly different (0.7 volts for the Parasound and 1.3 volts for the Sumo). But Julian didn't hear volume differences or channel imbalances. He reported quality differences in sound between the two.

The moral of this story is that a couple of dozen blind amplifier comparisons have pretty much proven that a modern, reasonably well-designed amplifier, with reasonably flat frequency response and operating within its power capacity, will sound just like any other well-designed amplifier. This latest test shows that amplifier differences apparent under the best possible long-term comparison conditions disappear when volume differences and channel imbalances are dialed out.

Becoming a geek is really fun. You don't have to chase those fragile differences that disappear when you close your eyes or the "coach" leaves the room. Now you can devote time, energy and money to system upgrades that can be demonstrated to others even when you're not there. Today that means better speakers, better recordings and DSP surround systems. Hey, man, it's freedom. Try it out.

* * *

Postscript by David Rich:

The experiences of Tom Nousaine in ABX testing are very similar to my own. I conducted my tests with dealers and manufacturers. Professionals in this field should, in theory, be able to make an accurate evaluation of the sound of a component. A custom ABX box was used for these tests. This box differs from the standard David Clark ABX box in some respects, one of which is self-contained level matching with a built-in set of passive attenuators. This one-off custom unit was built to tweako standards, including megabuck jacks, MIL-spec potentiometers, and Teflon wire. The ABX box was given to the test participants for use in the participants' own room with their own equipment. The participant was allowed as much time as necessary to become used to the ABX box and to find the most revealing program material.

At the start of the experiment all participants found the ABX box to have a minimal effect on the performance of their system. All participants felt confident that they could identify the differences between the equipment when the switch box was set to A or B. After the participants had worked with the ABX box for about a week, they would report that they were having more difficulty than expected in the blind test, but they remained confident they would eventually be able to pass the test. As time passed, the participants had to admit they could not achieve a statistically significant result. No participant agreed that this indicated that the components sounded identical; instead they blamed the ABX box for hiding the differences. They now claimed that the ABX box was so colored that differences could only be heard when the box was out of the system. They were unwilling to accept the evidence of their own ears. They had to find a scapegoat, and the ABX box was the only thing available even though they originally had no objection to it.

Nostalgia and Loudspeakers *(continued from page 33)*

rock artist to be emasculated, you can go out and get an Aphex Aural Exciter to add distortion back in, *so that it sounds loud again.* (Seriously.)

In summary.

If you are a true high-fidelity junkie (not an "audiophile"), if you have an insufferable doctor friend with Mcintosh equipment who needs to learn a lesson about the assumptions made by (mass-market) speaker manufacturers, or if you're simply tired of commercial offerings available to just anybody, you can have the peace of mind that comes from knowing that you have built something truly impressive, based largely on technology ignored for thirty years because of economic considerations.

*[Detailed construction diagrams are available from the author. Write to him c/o **The Audio Critic.**—Ed.]* •

Interviewing the Best Interviewees in Audio Part II

By David Ranada
Contributing Editor at Large

Here are two more of the acknowledged deep thinkers in the field, sharing with you their highly original and enlightened insights into the present state and future promise of audio technology.

Editor's Note: *It so happens that* ***The Audio Critic*** *has a history regarding both interviewees here. Bob Carver has been an editorial stormy petrel in these pages for years now, as most of our readers know. The color of his hat—is it good-guy white or bad-guy black or possibly pearl gray?—has been debated endlessly by audiophiles in general and our correspondents in particular. I think David Ranada manages to draw out the essence of the man in his interview, allowing Bob's tremendous enthusiasm and overflowing creativity to come through loud and clear.*

Mark Davis, on the other hand, is hardly ever mentioned in our articles, reviews, or correspondence, although he is certainly a brilliant practitioner in his own right. Long, long ago, however, he occupied center stage in just one letters-to-the-editor column, and the treatment he received there at my hands has been preying on my mind for the last few years, threatening to become a major guilt trip unless I expiate it right here and now. What happened was that Mark wrote to ***The Audio Critic*** *in 1977 that any two competently designed preamplifiers with identical frequency responses will be audibly indistinguishable from each other at matched levels. He had extensive research and experimental data under his belt to support that claim, and he complained that "I'm beginning to think it unfair that I should be the only one to have to carry the burden of proof." And how did I answer him? I made cruel fun of him, that's how. I just knew that the Mark Levinson had to sound better than the Dynaco, Yamaha, etc.*

So let's get this straight once and for all, Mark. You were right, and I was wrong. You were objective and uninfluenced by tweako belief systems; I was still thinking like a typical audiophile on many (though not all) subjects. I hope you have meanwhile forgiven me. More recent issues of ***The Audio Critic*** *explain in detail my current perception of reality, and today I refuse to defend some of the views expressed in those earliest issues. I still don't like cheap preamps but not because of the sound. (See also David Rich's comments on that subject in Issue No. 18.)*

5. Interview with Mark F. Davis, Audio Designer

RANADA: When I first met you, more than 15 years ago, you were already deeply into audio and psychoacoustics. How did you get that way?

DAVIS: I was into electronics as a kid.

RANADA: Just electronics or audio-related electronics specifically?

DAVIS: It started out as just electronics, I thought. I took a long electronics course in 10th, 11th, and 12th grade. But I guess I started getting into audio when I was five. At the age of five I can recall a little girl down the block having her phonograph break and asking me to fix it. And I remember taking it apart, not knowing what the hell I was doing. But it looked pretty with all the tubes.

RANADA: Why did she ask you to fix it?

DAVIS: Because I was interested in audio.

RANADA: Did you fix it?

DAVIS: No. At that point I didn't know that it was really broken. I think we were trying to figure out where the singers came out of. There was this little book called *Basic Electronics* that I took out of the library at one point. A lot of it had to do with audio because there just wasn't that much RF yet. My father had this enormous collection of old records, and in high school I would play them by just holding a phono cartridge in my hand and had the wires from the phono cartridge run to a crystal earphone—no amplifiers, very pure reproduction.

RANADA: And no de-emphasis either...

DAVIS: They used pre-emphasis in 78-rpm records?

RANADA: Yes, they did.

DAVIS: In this case, it was whatever you got was what you got.

RANADA: The case of a human tonearm, literally.

DAVIS: Exactly. Later on we had a Lafayette Radio in Syosset on Long Island. I used to go up there by bicycle on weekends and spend far too much money on stuff. I was very fascinated by tape recorders. I tried to build a tape recorder for a science project in sixth grade. It didn't work.

RANADA: Why was that?

DAVIS: Because I didn't realize that a tape head had to have two pole pieces that came around like a U magnet and came close together. I was trying to use an iron nail, which didn't work nearly as well. I couldn't find much written about tape recorders back then and I didn't know anybody who was particularly expert in them.

RANADA: When was this?

DAVIS: Sixth or seventh grade, around 1958. It's too bad I didn't live out here in Redwood City where Ampex was because I could have sat around their offices and asked questions. But this was all that Levittown had to offer.

RANADA: So you were destined for an electronics career at an early age?

DAVIS: So it would seem. I did see to it that I got into this electronics course in high school. And then I went to MIT and majored in electrical engineering and so forth. I tried to learn how to design circuits—because MIT doesn't like you to learn how to design circuits. They want you to learn the principles behind the designing of circuits. I have now come around to their point of view but at the time I wanted to learn how do design circuits.

RANADA: How did you get into psychoacoustics? There isn't necessarily a connection between electronics and psychoacoustics.

DAVIS: Absolutely. Through undergraduate years I had a strictly undergraduate curriculum, which had nothing specifically to do with audio.

RANADA: The basic EE kind of stuff.

DAVIS: Yeah, pretty much. I think there was this assistant associate professor, Barry Blesser, who gave a course on practical audio design. I think I took that.

RANADA: Did you ever take Bose's course?

DAVIS: Yeah, somewhere along the way I took Bose's loudspeaker design course. It was a very good course. Audio was still my hobby, and there was this local hi-fi shop—Tech Hi-Fi—that I worked at part-time as a salesman and so on. But it was strictly a hobby. I was meandering down the hall one day with my master's thesis supervisor-to-be, Campbell Searle. He brought up some problems he was having with his hi-fi system, and we got into a long, involved conversation about his hi-fi. And I somewhere sort of brushed off audio as this silly side-hobby that one doesn't really talk about in the halls of MIT. Campbell said, "No, no, no, it's a perfectly serious and reasonable area of study" and so on, and that I should take it seriously. That was at lunchtime. I went home and over lunch hour I typed up a four-page critique of his hi-fi system and brought it back to him. He had made some Klipschorns at home. When Klipschorns had first come out, they were very popular. There had been a little group of MIT students—this was way before my time—who had made Klipschorns, so I think he still had a couple of them. And he was biamping them with these little Heathkit amplifiers, with 30 watts per channel or something like that.

RANADA: That should be enough for a Klipschorn.

DAVIS: And they were biamped, so they could really put out a lot of sound. But I think I had critiques about how he was setting the relative levels of his tweeters and his woofers, and how he was doing the crossovers and things like that. That sort of started a more-or-less serious association between Campbell and me, exploring audio.

RANADA: He was a psychoacoustician?

DAVIS: He had been a straight circuit-theory guy. He and Paul Gray, who for a time was a president of MIT, had written a book on circuit theory, a great big book that was known to everyone as "Gray and Searle." Having done that, he reached a point where it was evident to him that circuitry per se was going to become very "cookbook," and something that only chip designers worried about, where you manipulate the stuff using computer circuit-design programs. Since it [circuit design] was clearly drying out as a major area of research, he decided that he needed to shift his academic focus and proceeded to go into psychoacoustics, where the intent was to apply the sort of signals-and-systems theory that had evolved in the electronic disciplines to human hearing—to treat [hearing] as a signal-processing system and then try to characterize it as best as possible to try to establish causative relationships between what sound went in your ears and what the resulting perception was.

RANADA: This is a very different way of looking at perception, as sort of an analogue of an electrical circuit.

"Each person's pinna characteristics were very individual.... If we gave somebody a new set of pinnas, in some amount of time they'd ' learn' [how to use] them."

DAVIS: Exactly. You are not caring primarily about the biological functions or anything like that. You want to know how the whole thing acts as a system. And although tracing things down to basic physiology is a useful endeavor and a check to make sure that your model of the system is accurate, it's the models that evolve that are really important. So for my Ph.D. I got seriously into psychoacoustics, and there was a very strong combination of applying electrical engineering and psychoacoustics. I was using computer-based DSP to try to create signals that would impart the illusion of a sound coming from a predetermined direction [a phenomenon called "localization"]. I was kind of knee-deep in the tying together of EE, signal processing, and perception.

RANADA: Let's go into this a bit. I remember being an experimental subject of yours and you pouring gook into my ears in an effort to make molds of my outer ears. I assume that this was part of your dissertation.

DAVIS: That's right. That actually preceded the formal dissertation. That was exploring to what extent the pinnas imparted localization. And one of the things that was done was that we made these gooey models of all of our ears—first negatives and then positives—and made recordings with microphones in a dummy head with these ears fitted to the outside of the head. We found that you could make recordings that did impart a greater amount of spatial orientation than if you just used a plain-vanilla KEMAR head [a standard psychoacoustical manikin]. When you tried to play one person's recordings to another person [we found] that it didn't work generally very well. Each person's pinna characteristics were very individual.

RANADA: Can you explain how the outside of your ear affects what's going on at the eardrum?

DAVIS: As far as we can tell, the little curves on the ear cause little reflections that affect the frequency range from about 5 or 6 kHz out to maybe 14 or 15 kHz. The effect of these [reflections], depending on the direction of arrival of the sound, tends to impart features like deep, narrow notches at various frequencies in the spectrum [as received at the eardrum]. And as the sound moves around, the notches move around [change frequency], and your brain learns the pattern of these notches and how they correspond to directions.

RANADA: So that explains why one person's ears won't work on another person, because the notches are in the wrong place.

DAVIS: That's right. What notch goes where is a function of how the sound moves around. In time, you sort of learn your own ears because you have visual and other feedback to tell you [which direction] the notches correspond to, and so on. You're constantly moving your head around and often looking at sounds that you care about and subconsciously—but actively—tracking the notches. If we gave somebody a new set of pinnas, in some amount of time they'd "learn" them.

RANADA: What would happen if you gave somebody their own pinnas but when they were five years old and much smaller?

DAVIS: That's an interesting question.

RANADA: I would assume that the relative positions of the notches would remain where they were but that their frequencies would change.

DAVIS: Right—things would sort of slowly lower [in frequency with age] but basic shape of the variations with position would remain the same.

RANADA: And you'd be able to tell whether it's the relative positioning that was important or whether it's the absolute frequency that matters.

DAVIS: That's a very interesting suggestion. I don't ever remember anybody trying that. You have raised what I think is still an outstanding unknown of research about all this, namely what is the important information. Is it the notches, their depth, or their width, or their center frequency or what?

RANADA: Along with all the psychoacoustics, you had to learn the basic mathematical and statistical principles for evaluating how things are.

DAVIS: That was a major stumbling

block. I knew signal theory going into the thesis and I knew something about hearing but I didn't know a whole lot about statistical analysis. And a great deal of time had to be spent on just coming up to speed on that.

RANADA: There are a lot of people who complain about controlled listening test for all sorts of reasons. But they always complain about the statistics' being not meaningful. What are your thoughts about that?

DAVIS: There are admittedly some conditions that apply to just casual listening that don't quite get faithfully reproduced in listening tests. The usual complaint is the long-term listener fatigue kind of thing, where listening tests are generally relatively short in duration. Listener fatigue is something that is very difficult to measure under any conditions because it's tied in not just with the sound but the listener's physical condition at the time. You've got a lot of variables that are difficult to pin down. In general, I think that controlled listening tests—done by people who have some experience in conducting them, [as well as] an understanding of what the underlying mechanisms being tested are—can be pretty valid and pretty revealing. If you can get a multiplicity of testers to agree on a testing paradigm and to agree to administer a test in similar fashion, you can get usually pretty similar results across a pretty broad spectrum of subjects [listeners]. We all do have human ears and there are pretty well-established norms as to what those ears are and are not capable of doing.

RANADA: Your dissertation topic was still some distance from the hi-fi world. How did you get from MIT into audio?

DAVIS: The Boston Audio Society is the major culprit in that. I started going to their meetings and interacting with people who are actually in the hi-fi biz. I would go and listen to these people give presentations on their hot new products. In some cases it was very impressive, but in other cases I'd find myself sitting there thinking, "This is silly—I can do better than that." So I actually started messing with designing hi-fi circuits well during the time I was in college. I got interested in noise reduction and built noise-reduction systems—wideband companders and things like that.

RANADA: Just for your own amusement?

DAVIS: Just to see what they did. I briefly considered going into business and making them and decided that was not what I wanted to do. I did phono preamps, trying various and sundry FET and bipolar transistor phono preamps, and loudspeaker equalization and stuff. In college I got really interested in what really made one loudspeaker sound different from another or whether they sound different at all. The lab at MIT had this 31-band graphic equalizer with very steep filters. It cost several thousand bucks at the time. I would drag that home with B&K calibrated microphones and related equipment. We'd set up a whole bunch of loudspeakers and try to equalize each of them very, very carefully to the same V3-octave response four feet in front of them. We'd listen to see whether we could hear the difference between loudspeakers or not. We found when speakers had similar radiation patterns and when they could produce similar acoustical output [volume levels], you could make them sound astonishingly similar. The larger, more expensive speakers could deliver more acoustic output, so there were times when I could make a very cheap speaker sound like a more expensive speaker as long as I didn't turn the volume too far up. Once you were past a certain point, the little one would start "flapping." We also found that if the radiation patters were materially different, there was no way you could make one speaker sound like another under any conditions. But it wasn't that hard to imitate the radiation pattern. At one point I recall comparing an AR LST loudspeaker, which was a speaker with a front-facing panel and two diagonally outward-facing panels, with drivers on all three. We attempted to equalize an AR-7—one of their smallest [two-way] speakers—to have the same response, and we found that when we did they did not sound the same. But when we took two AR-7s, one on top of the other, and canted them a little bit so that they were pointed in slightly different directions to imitate the slightly different directions of the AR LST drivers, then we found we could produce an extremely good match between those two. That started me thinking about what would be good loudspeaker radiation patterns and so on.

"We found when speakers had similar radiation patterns and when they could produce similar acoustical output [levels], you could [by equalizing them] make them sound astonishingly similar."

RANADA: What was your first product to hit the unsuspecting consumer?

DAVIS: That was the Allison Electronic Subwoofer. But I think that was preceded by a little phono preamp—the Davis-Brinton phono preamp—though it was not generally offered for sale. Jim Brinton [former president of the Boston Audio Society] and I did that. Jim was friends with Roy Allison and sort of got us the contract to do the design for the electronic subwoofer. That [product] was indeed sold, albeit in small numbers. Out of the noise-reduction work that I did came a little compander built around a Signetics compander chip. Joel Cohen used that circuit to noise-reduce his bucket-brigade delay line for the Sound Concepts ST-550 [an early ambience enhancer]. Toward the end of my college career, I got hooked up with Francis Daniel in New York, who had started Benchmark Acoustics, and largely on the basis of Francis' well-tuned ear we put together an ambience box that was called "The Other Half." It was vaguely like the Sound Concepts in that you would take the signal and pass it through this thing, and then it would either feed front speakers and/or side speakers. It was a much more subtle sort of effect than previous ambience-type units. Francis was trying to get rid of the harshness that tended to accompany ordinary stereo reproduction, particularly on direct-firing speakers.

RANADA: I have always wondered why some stereo systems sound harsh when played loudly.

DAVIS: Because of the fact that the upper midrange and high-frequency sounds are being beamed very narrowly at the listener and not around the room, whereas the rest of the sounds—due to the radiation pattern of the speaker—are getting bounced around the room. The middle and high frequencies are like a pair of headlights shining in your eyes, whereas the rest of the spectrum is more like diffuse background lighting.

RANADA: Why does this sound worse when it gets louder?

DAVIS: I'm not entirely sure. Partially probably because the effect falls below thresholds. In any case, even at lower levels it can add an unnatural amount of false brightness. What Francis' observations led to was that you had to try to make, on the one hand, the radiation of the energy a little bit more diffuse, but at the same time you had to do it in such a way that the imaging did not also become more diffuse. So evolved "The Other Half," which did a nice job of easing the problem of making loud passages much less harsh without either rolling off the treble or compromising the imaging particularly. I finished the design of that one and Francis went about trying to get it produced and promoted. In the meantime I started on the design of a loudspeaker intended for Benchmark Acoustics. But before I got a little way into it, it became evident that that project was going to be a good deal larger, more expensive, and more involved than poor little struggling Benchmark could afford at that point. So we arranged that dbx would buy the rights for the speaker. I went over to dbx to work on this speaker and to work on noise-reduction systems.

RANADA: You do seem to have had the perfect background to work there.

DAVIS: It did seem to kind of all come together with all the systems theory, the electronics, the psychoacoustics, and the real-world experience designing stuff.

RANADA: Could go into the psychoacoustic background of your speaker?

DAVIS: There were a couple of fundamental considerations that went into the [first] dbx speaker. One was the notion of using cross-firing systems to try to widen the imaging space [the listening area where you can get a good stereo image]. That, in and of itself, was not unique to dbx. [The idea] had been floating around in hi-fi circles for a while—you could improve the imaging area by cross-firing whatever speakers you had slightly [that is, aiming them so that their front axes cross in front of the listening position, not at it]. But what happened was, we started looking at how well that worked and how to make it work as well as it could. We found, not very surprisingly, that a constant radiation pattern was needed. As you moved around the speakers, the balance between the left and right speakers had to vary according to a specific characteristic. We could measure this characteristic readily enough by moving people around, having them stand still, and adjusting left/right levels [so that the image still seemed come from between the speakers].

RANADA: So this experiment took in both level and timing differences.

DAVIS: It took them into account in that the timing differences were present, and the listeners would turn the left/right balance so that the image was most central and the timing [differences] went into their judgment. At that point, not only were we interested in measuring what the settings were, we were also interested in how good that image could be. We found that when using conventional speakers to do this measurement, as long as we kept the speakers pointed at the listener so that the frequency response did not change, we could obtain astonishingly good imaging at distances way off axis. From this comes the notion of a constant radiation pattern.

RANADA: When you say constant radiation pattern, you mean over frequency?

DAVIS: That is right, it's not over angle. [What you want is a frequency response that does not change with direction of radiation.] That also came about from the earlier experiments showing that nonconstant radiation patterns led to harsh high-frequency response. It was also my observation that typically, on one of these direct-firing loudspeakers, the sound appears to come from somewhere near the speaker, except for the high frequencies, which seem to come directly from the tweeter. I felt that by doing a constant radiation pattern you would get rid of this clue that you were listening to a tweeter, and that the high frequencies would be better integrated with the entire rest of the image. The point, and the difficultly, was that if you did these experiments, what you came up with was a radiation pattern that, as you walked around the speaker, had to get louder at some points and softer at others. But it had to do that at all frequencies by about the same amount, at least above 200 to 300 hertz, below which point there's nothing really localizable. Since real-world drivers simply do not have constant radiation patterns, we set about trying to make a system with a combination of drivers individually contoured so that their net acoustic output was this constant radiation pattern. [The frequency response in any direction was the same, but the overall loudness of sound radiated in that direction had to change by a specified amount.]

RANADA: This was similar in principle to phased-array radars, then.

DAVIS: That's right. What resulted was this enormously complicated crossover network that got put into the base of the speaker, and fourteen drivers (four woofs, four mids, and six tweets), and also an outboard equalizer because the aggregate frequency response of the system, particularly with all this crossover stuff, was far from flat in and of itself. In subsequent models of the dbx speaker we got better at doing it with fewer drivers and we concentrated on front-firing [radiation], whereas the original one was more omnidirectional. Eventually I did an [acoustic] lens-based system that got by with just a couple of drivers, but we never got that into production.

RANADA: Your acoustical lens was made out of foam? Plastic?

DAVIS: It was sort of built out of hunks of wood, a little plaster of pans, and a few socks stuck into the grille holes at appropriate places.

RANADA: You don't recall the brand of sock, do you?

DAVIS: No, I don't.

RANADA: This would make a great construction project for a magazine.

DAVIS: Yeah, it actually would. I kind of would like to go back and build one of these systems myself because the imaging was really excellent; it was superb. It used just a pair of coaxially mounted drivers with an individual acoustic lens for each one, and it produced very, very good imaging. One of these days...

"...as long as we kept the speakers pointed at the listener so that the frequency response did not change, we could obtain astonishingly good imaging at distances way off axis."

RANADA: That was one of your prongs of research at dbx. The other was noise reduction. You have the unique honor of being the only name on the patent for the [United States'] stereo TV noise reduction system.

DAVIS: At that time nobody knew if anybody really wanted stereo sound for TV. They still didn't have widespread stereo for AM radio; nobody seemed to want it. So I puttered away on that [dbx's] system. I got to bring to bear a lot of the experience and a few of the prejudices that I had developed in college in playing with noise reduction.

RANADA: What were some of those prejudices?

DAVIS: At the time, there were two classes of noise reduction around: the dbx and the Dolby. The dbx was a wideband system that could do enormous amounts of noise reduction under the proper conditions, but there were times when it would be imperfect and you could hear the noise—and the fact that it [the noise] was [audible] only at times was very obtrusive. The Dolby was a variable-filter kind of noise reduction system; it was much more conservative. It worked primarily at high frequencies, which is where the primary noise components were. Although it did not totally eliminate the noise, it was much less obtrusive and seemed to reduce the noise by a constant amount. So one of the prejudices was—I felt that combining those two systems and using the best points of them was a viable way to go. It was also becoming practical because the cost of high-performance VCAs [voltage-controlled amplifiers] was coming down, dbx now had theirs in a chip form instead of a circuit module. There were three key tests I did at dbx that really specified the system. The first thing I did was measure the TV-audio channel as a function of frequency for its overall [frequency-response] characteristics and its noise floor. This gives you two curves you can plot on the same graph. One was the maximum signal you could put in as a function of frequency, which was like a flat line that rolled off at higher frequencies. The other was the noise floor that started maybe 50 dB below. [This curve] was more or less a flat line, but it came up at higher frequencies. You had less overload [margin] and more noise at higher frequencies. At midband you may have 50 dB of signal-to-noise ratio; at high frequencies you may only have 30 dB. I seem to have been the first one to analyze the problem in this way, [in terms of] dynamic range as a function of frequency. Prior noise reduction systems pretty much dealt with the channel as having a single overload point (independent of frequency) and a single aggregate amount of noise (independent of frequency). Masking is the operative psychoacoustic principle that you are exploiting in noise reduction. But even though it was acknowledged that masking is frequency-dependent—[leading you to] a frequency-dependent system like Dolby A—it was a semiarbitrary decision as to how much noise reduction was applied. Here I was saying, "Let's measure exactly how much noise reduction you need and provide a system that provides exactly that much and no more." If I have a channel that has 50 dB of dynamic range at midband and 30 dB at high frequencies, and I needed 90 to 100 dB when it's in noise-reduced mode, then I derive from that that I need 2:1 compression from the midfrequencies on down, and at high frequencies I need about 3:1. Well 3:1 is a lot of compression—even 2:1 is—but that was the first time that somebody said you've got to bite the bullet if you really want to have a chance at getting this signal through the channel. So that was one measurement: the overload point and the noise floor. The second measurement was a characterization of the class of signals that would be going through each channel, which consisted of me watching a real-time 31-band spectrum analyzer (an Eventide unit that was plugged into an Apple II computer) for a wide variety of material—speech, music and so on.

RANADA: You didn't take any data; you just looked at it?

DAVIS: For long periods of time. I tried to figure out how I would try to get this moving bar graph of sound through this rather narrow channel. What came out of that was a characterization of typical audio signals as consisting of relatively strong mid-to-low-frequency fundamentals and a rolling off series of harmonics above 1 kHz or so. What varied, in a first-order approximation, was the overall height of this funny mountain and the slope of the high frequencies. If you could control the up-and-down motion of the whole thing and you could control the [high-frequency] slope, you could fit the resulting spectrum through the channel most of the time. What came out of this was the notion of combining a wideband compressor with a variable high-frequency filter, something like putting a dbx and a Dolby B together. But the characteristics of that high-frequency filter were definitely not the characteristics of a Dolby B filter, which is a sliding shelf. The characterization that came out of watching the signals on the analyzer was that you needed something that would more or less hinge around 1 kHz and have more effect at higher frequencies because you had progressively narrow dynamic range. The third thing was, what should happen in the absence of high frequencies in the program material? How sharp a filter do you need? Here's the case where the preemphasis [treble boost] on the encoding side is at its maximum and the deemphasis [treble cut] is maximized on the playback side. How sharp a filter do you need to get rid of the noise so you won't hear it? I did a simple test: I put a 400 Hz tone into the electronic simulator of a TV-audio channel and adjusted a filter until I couldn't hear the noise anymore. It turned out that the filter had to be a second-order (12 dB per octave) filter in order to get rid of the noise. This was important because that is the sort of thing you need to get a piano to come through without "breathing." [If] somebody hits a piano note, which is mostly midrange, you don't want the high-frequency filter to "open up" [and let through all the noise]. My funny little variable filter that I was going to tack on to the wideband compressor had to do a 12-dB-per-octave boost or cut. But it also had to go down to no boost or cut when the signal had a lot of high frequencies. It needed this range from no boost to a 12-dB-per-octave boost, which was unusual for a noise-reduction filter. The noise-reduction filters before, and most of them since, are pretty much 6-dB-per-octave animals. So this was a sharper filter than had been used before. Anyway, those three things all put together, along with a limiter that was put inside the noise-reduction loop so that you wouldn't hear its effect, became dbx's entry against CBS Labs' and Dolby Labs' entries for the selection of the TV NR system.

RANADA: One of the accomplishments of your system is that stereo TV is actually quieter than mono TV—is that correct?

DAVIS: No, it's no noisier. The stereo TV system uses basically the same kind of signal flow as in FM stereo. You have a main audio channel, which is driven from the sum of the two stereo channels. On a second [transmitted] channel, which is decoded by the stereo decoder, you send the [stereo] difference information, the left-minus-right signal. Then by alternately adding and subtracting those signals in playback you can recover left alone and right alone. In both the TV and the FM stereo systems, the L-R channel is in fact a good deal noisier than the L+R channel. So that if you just listen to the L+R, it's generally pretty quiet. But as in FM, when you put in the L-R and try to listen in stereo, it will generally get at least 15 dB noisier. However, by using this fairly aggressive noise-reduction system on the TV L-R signal, the resulting noise is actually lower, in most conditions of use, than the noise on the main channel, which is not noise-reduced. (As the main channel was already in use, we couldn't add noise reduction to it, since then it would not have been backward compatible.) So we left the main channel alone and we applied the noise reduction only to the difference channel. The difference channel [when] noise-reduced is quieter by a good amount than the main channel, so in switching from mono to stereo things don't get any noisier. It's too bad that we didn't have things like that around when we did FM stereo.

RANADA: Soon after your TV stereo NR system was indeed selected as the best submission, you were hired by Dolby and

"It's not that you are necessarily throwing relevant information away; it's just that you are only sending information that the ear is responding to and not extra, redundant information."

have been working on digital approaches to noise reduction and signal coding. I wish you could explain how the new digital-audio data-reduction schemes can actually work, since you are throwing out much of the data.

DAVIS: This gets into audio coding, which is kind of the DSP equivalent of analog noise reduction.

RANADA: But in noise reduction you aren't throwing away any of the signal, are you?

DAVIS: In a sense you are, in that when the signal gets into the channel you have all this noise that the channel adds that wipes out any part of the signal that is equal to or below it [the noise]. On a simplistic level, when you digitize something, each additional bit of accuracy is equivalent to 6 dB of audio dynamic range. So if I start with 16-bit PCM samples, multiplying 6 dB per bit by 16 bits gives a total theoretical dynamic range of somewhere around 96 dB. In digital coding you are throwing away bits, and if I try to throw away ¾ of those bits, if I have only 4 bits per sample, that's 24 dB [of dynamic range]. [This 4:1 compression] is like I'm trying to compress a signal with a 96-dB dynamic range through a channel with a 24-dB dynamic range. Many of the same psychoacoustic principles apply and many of the same or similar techniques apply. But where an analog noise reduction system might have two or three or four bands or something like that, a DSP system can have 40 bands or so. This enables us to effectively deal with [transmission] channels that are much noisier than anything we have dealt with in the past. This has been a very exciting area and as the world increasingly carries audio information around in digital form, coding [as this process is called] is becoming an increasingly important element.

RANADA: There are people who say, "How can you possibly preserve anything when you are getting rid of all of this data?" Why wouldn't the ear be able to detect that?

DAVIS: Basically what the coders do is, they transform the signal into the domain that the ear is using and then they send only the actual information that the ear will respond to. For example, if I put a 1-kHz sine wave through a simple 16-bit PCM system with 44.1-kHz sampling (as in the Compact Disc system), the fact that I have a 1-kHz sine wave is sort of irrelevant. The system will send 44,100 samples per second times 16 bits per sample, some 700,000-odd bits per second [705,600 to be precise]. That's how much I'm sending, regardless of whether there's a 1-kHz tone there or nothing or a symphony orchestra. But the ear is going to hear the 1-kHz tone as an isolated tone. So we do a Fourier transform [spectrum analysis], and what comes out of that is all these narrow frequency bands, and all of them are zero except the one for 1 kHz. It takes very little information to send to the decoder the fact that all the bands are zero except for the one at 1 kHz. It's not that you are necessarily throwing relevant information away; it's just that you are only sending information that the ear is responding to and not extra, redundant information. Systems without this kind of processing often get into the situation where they are sending completely redundant information that's many, many bits but very little audio. I guess the worst case would be if I put nothing—utter silence—in. One of these plain-vanilla systems will still send 44,100 16-bit samples per second—all being zeros. A coder wouldn't do that; a coder would send "everything is zero"—one little message. There wouldn't be any need to send anything more on the channel until something actually happened. But you're not losing anything in that case; there's no implicit distortion of the audio. If what you put in is 1 kHz, what comes out—if it's properly coded—is that 1 kHz. For the most aggressive coders we actually go beyond that. We are actually throwing away information. While it's not audible, it's at least measurable. My first point was that you don't even have to throw away anything that's measurable; simply express the information succinctly, and you can send fewer bits. But we do get into cases where we are throwing away enough bits that the differences are measurable by sufficiently sensitive instruments. However, the differences are meant not to be audible. What this involves is going back to psychoacoustics and using somewhat more elaborate models that predict what parts of the signal are audible and what are not. For example, in the case of my 1-kHz tone, suppose I had an additional tone at 1050 Hz that was 50 dB quieter. I could pretty much skip transmitting anything about the presence of that 1050-Hz tone because you simply won't hear it next to the 1000-Hz tone. If the 1000-Hz tone goes away, *then* you do have to send the 1050-Hz tone because it [the 1050-Hz tone] is no longer being masked. So we do make use of masking to throw away information. But the net result is that you can still do an A/B listening test of your original signal against what comes out, and they should sound identical. [PASC, the 4:1 digital coder used in] DCC is by no means the only audio coder that will become a widely held standard, although it may become one too. One of the other ones that seems to be bubbling around is a standard for computer-based storage of music and interactive multimedia playback. There is a working group, with about 170 corporations participating, that is defining standards for coders that will run on PCs and that will provide bit reduction on the fly on PCs in an interchangeable format across [computer] platforms. These coders, because they actually run on the CPU and do not make use of a formal DSP chip, are much less elaborate than the sort of coders that are being put into DCC. These standards are being evolved now, and you'll see them in the next year or so.

RANADA: These aren't going to claim that you can get CD-quality sound, are they?

DAVIS: Right. The question is how close can you come—can you make the coder unobtrusive?

RANADA: I want to get into the philosophical issue we covered once in another conversation: whether you think perfect high fidelity is possible at all, whether simply trying to create in the listener's brain the same perceptions of something that has occurred somewhere else at another time violates any physical laws?

DAVIS: There's this theorem by Laplace that says *effectively*—imagine yourself sitting in a concert hall, and you surround yourself with an imaginary array of an infinite number of microphones located on this big sphere surrounding you. Each one is connected to a loudspeaker just on the other side of this imaginary spherical membrane. You convey the sound through the membrane via a zillion little microphones on one side and a zillion little loudspeakers on the other. If you had an infinite number of them, in theory you would have a perfect sound conveyance system.

RANADA: Is this a mathematical theorem?

DAVIS: Yes, the basic theorem operates within a region—in any wave-bearing medium, it doesn't have to be sound waves in air—in which there are no sources. Assuming that you have wave energy arriving from strictly outside this sphere—there are no internal sources of wave energy—then the pattern of waves within the space is entirely specified by just knowing the normal [perpendicular] component of the impinging waves on the surface of the space.

RANADA: Now this would seem to be one of those mathematical theorems that would require, for perfect reproduction, an infinite number of everything.

DAVIS: Right, in and of itself, it would require an infinite number of everything and would also mean that you would end up with an infinite bandwidth. Potentially, if you fulfilled the strict requirements of this theorem, you would need an infinite bandwidth [for mikes, loudspeakers, and any recording medium].

RANADA: You'd be able to put your dog inside the sphere, and it would respond to a dog whistle outside it?

DAVIS: Exactly. So we very quickly get to the point of saying that you can get by with fewer than an infinite number of these things and still not hear any difference—and still have a perceptually perfect system. Then the question becomes how far below infinity you can take it. You can't hear anything above 20 kHz, and the wavelength of 20 kHz in air is something like ¾ inch or so. I think, in all likelihood, if I have a microphone/speaker pair only every ½ inch around the sphere, that that would probably be enough. If you work that through and you still wind up with a god-awfully high number of channels, then it's still not practical at that point.

"In theory, you should be able to encode a three-dimensional [coherent] sound field sampled at three or four dozen points into a [reasonably low] data rate."

RANADA: This would be hundreds or thousands of channels? This would, I take it, depend on the size of the sphere.

DAVIS: This is true too. I suppose it would not have to be any bigger than your head. So that might not be too bad, having ½-inch resolution around a sphere with a one-foot diameter; I'd have to work that out but it'll probably come to the hundreds of channels.

RANADA: So you'd be putting on your stereo helmet...

DAVIS: Of course, you don't want to have to wear a stereo helmet. If you want to move your playback transducers back to the walls around your room, then that becomes the theoretical boundary and you might need a lot more channels in that case. You know that in signal processing there is the digital sampling theorem that says you don't have to sample at a rate more than twice the highest frequency contained in your signal. There's a kind of perceptual corollary to that regarding [this kind of] spatial sampling. Each one of these channels is basically a sample in space of the sound at that point. The ear can only sense a certain minimum change in angle. It varies as a function of position and direction, but for horizontal directions—say for a sound right in front of you moving left to right horizontally—the just noticeable difference is on the order of about one degree of arc. If the source was directly in front of you and moving up and down, instead of back and forth, the minimum difference would be about five degrees of arc. [The sound source] would have to move that much before you could clearly hear a change in position. Just from these specifications, if you chopped up the solid space around a person into one-degree segments in the horizontal direction and five-degree segments in the vertical direction, you would again get a few thousand channels. But that's still quite a bit fewer than if you had a speaker every ½ inch around your walls. And so that is a first, simplistic cut at applying what might be considered a spatial sampling theorem to reduce the number of channels.

RANADA: Now, the object of this exercise is to create around the listener's head the exact same sound field that he would have experienced in the original recording environment?

DAVIS: Yes, to the extent that the listener can perceive it as being different or the same. Now there's another thing you can use which is certainly used in present audio systems, and that is phantom imaging. In fact, I can have a pair of sound sources farther apart than one degree and still create the illusion of sounds coming from the space between those speakers, by adjusting the timing and amplitude of the sounds coming from them. And that's basically all we've got in two-channel stereo. It's pretty crude. But if you had a speaker every 5 degrees of arc horizontally and every 15 degrees vertically, that would sound pretty close to perfect.

RANADA: Now these are just simple systems, just microphones and loudspeakers—no other processing?

DAVIS: Right. Where the processing comes in is that, even with only a few dozen channels, the storage requirements are enormous and beyond what existing storage media can comfortably handle if you are going to make a digital recording. And [they are] also hopelessly beyond what the ear can possibly absorb. Assume that three dozen channels would somehow cut it. You've got 36 channels, each running at 700 kilobits per second—you're talking about oodles and oodles of bits per second. You're talking 25 megabits per second. There's no way that the human auditory system can possibly absorb 25 or so megabits per second.

RANADA: We talked once about what the maximum data output rate of the human ear is; it's well below that figure.

DAVIS: In theory, you should be able to encode a three-dimensional sound field sampled at three or four dozen points into a much, much lower data rate. [But without the employment of special coding techniques] effectively what you are coding [in this case] is three or four dozen channels, each playing a totally different program. There is no way a human listener could absorb all those programs simultaneously. All that is going to come out is this horrible cacophony, and nobody will understand a thing. We don't want or need a system that can play 36 different, simultaneous, discrete, independent programs to a listener. What we want is a system that will transmit faithfully a three-dimensional, coherent sound field that is one ongoing sonic event. It is clear that we ought to be able to apply some sort of relevant coding to [compress] this entire sound field down to some sort of reasonable number of bits, commensurate with what the human ear can actually absorb. This is a very active area of investigation. You [should be able to record] on a standard Compact Disc, instead of just two discrete channels, a complete three-dimensional sound-field event—in coded form—that could then be reproduced with arbitrary accuracy. If you've only got a mono speaker, then what can you do but play it through that speaker. If you've got two speakers, you'll get stereo. If you happen to have taken the time and trouble to set up 36 speakers in your room, this thing will decode the 36 channels and—in theory—you should have, within perceptual limits, a perfect recreation of the original recording.

RANADA: Thirty-six speakers is still a lot of speakers. Is any research being directed to make such a system more practical?

DAVIS: We're still not at 36 in terms of any real-world systems but we are at six, and that is the Dolby AC-3 coding technology, among others, which is being used in Dolby SR·D films. The AC-3 coder is my little invention; I'm the principal inventor. The Dolby AC-3 coding technology is derived from Dolby's AC-2 coding technology. AC-2 technology is one-channel-at-a-time coding using a filter bank based on a running transform. The description of a 40-band system I gave is based on AC-2, and that is the starting point for AC-3, but AC-3 takes in multiple channels at a time. (With the current implementation there are six; they are intended to be left front, center front, right front, left surround, right surround, and a limited-bandwidth subwoofer channel.) The AC-3 coder takes running transforms of each of those channels and then encodes the entire mess of transform coefficients in such a way as to use a minimum data rate—at least for the state of the art at the moment—without compromising fidelity or introducing coder artifacts, while using fewer bits than would be possible by using six channels of AC-2 coding. In fact, we save about half the bits that way. By specially coding redundancies that exist across channels—by knowing, going in, that because you've got a coherent sound presentation there will be significant redundancy—we can acutally encode those redundancies in a way that doesn't impair the sound but does reduce the required bit rate. And, of course, this system is being actively used now to encode movie soundtracks. The ingoing bit rate is six channels going at 48,000 16-bit samples per second [4,608,000 bits per second total]. (The system will shortly be using 18-bit converters which will increase the incoming bit rate but not the coded bit rate.) The coded bit rate for the audio is 325,000 bits per second for all six channels.

RANADA: That's less than a quarter of the standard CD bit rate!

DAVIS: That is quite correct; it is a data-rate reduction by more than a factor of 12, from 16 bits per sample to about 1.25 bits per sample.

RANADA: PASC used in the DCC system gives only a factor of 4 reduction in bit rate compared to a CD. Using AC-3 technology at the normal CD data rate, then, you could have 24 channels?

DAVIS: You can see that it is now practical at this point to code it [onto CD]. Tomorrow afternoon we could make a CD that had 24 channels of [sonically related] stuff on it if we wanted.

RANADA: Has anybody done experiments using standard audio recordings, as opposed to movies, to see whether this perfect-reconstruction theory is the right direction to go?

DAVIS: We have obtained six-channel mixes of various and sundry audio recordings, jazz combos, and what have you. And they come through just fine, thank you.

"You can see that it is now practical...to code it [24 channels on a CD]. Tomorrow afternoon we could make a CD that had 24 channels of [sonically related] stuff on it if we wanted."

RANADA: The trick is now to combine this with some sort of sound pickup technique to start approximating the infinite sphere of microphones.

DAVIS: There's two parts to this, I think. One is just pickup techniques that can take advantage of having lots of channels. Since commercial recordings are often made in layers and stuff—you put up four microphones to mike this part of the orchestra and four others to do this and so on, and you array those in various channels—I think some people might just want to make 24- or 36-channel recordings. I think eventually you will want to have honest-to-god microphone "trees" with an awful lot of microphones on them. I don't think the microphone companies are going to mind selling groups of 24 microphones at a pop. I think synthesizers will start to be multichannel in their orientation and the control that they will allow. You will have little mouse pads and things like that to control the trajectories of sounds and so on. I think it would open up music composition tremendously; I think it would be a real revolution, from an artistic point of view, to give this kind of control to the artists: to enable them to put any sound they want anywhere in space they want and have it reliably come out at the end.

RANADA: So at least in the number of channels available, we are approaching the level of technology needed to transmit to the home everything needed for perceptually perfect "3-D" sound reproduction. The question is: how can it be made practical in the home, where getting only two speakers installed is often a problem?

DAVIS: I think if you really want to be something like perfect, you're going to be looking at a few dozen loudspeakers.

RANADA: There's no way to get around that using very clever psychoacoustic trickery?

DAVIS: Not yet; there's only a finite distance that you can count on phantom images to be pretty solid. You can't get too far afield before the room begins to impose its character. If you want to dominate the room characteristic, you're going to need a lot of loudspeakers. But I think that this is going to be very strongly a matter of personal taste. I think that for a lot of people by the time they have, say, five speakers around them in a horizontal arc and maybe one on the ceiling, that a lot of them are going to feel quite happy. Our experience to date in simply listening to movies produced in six-channel discrete compared to the previously available Dolby 4:2:4 matrix system is that a significantly enhanced perception of sonic reality is imparted by just going from the four matrixed channels to six discrete ones. That's already a big step. For a lot of people the approximation to reality will be sufficiently close by the time they get a half dozen speakers. I do think that there will be audio equipment that comes out that will help support this [multichannel effort]. For example, if you've got 24 speakers, you're going to want 24 amplifiers, but none of them individually has to be particularly powerful. So you could see receivers with 24 amplifiers in them, each of which would have to put out maybe only five watts. It's not that hard to do.

RANADA: All this sounds tantalizingly close. But to be a commercial success, there would probably have to be some standardization to prevent market chaos. Do you know of anybody working toward such standardization of multichannel coders?

DAVIS: Standardization is a very big issue. At least with our Dolby AC-3 technology, we are working toward trying to establish that as a standard. Certainly, if it is used in motion pictures it would be to the benefit of the motion picture industry to adopt it as a worldwide standard. As it is, 35-mm films are a worldwide standard; you can take any 35-mm print and play it anywhere. It is highly desired to continue that. It's one of the reasons we put the digital AC-3 sound track in between the perf holes on the film[!], so that it wouldn't disturb the analog sound track or the picture, so that everything is compatible. In addition, I think I can reveal that there are already one or more chip makers who are working on building a cheap single-chip decoder for this system. Starting from the movies, this system could become a standard for broadcast use, since broadcasts often show movies. Why not just take the bit patterns that are on the film and send them digitally? This could be done tomorrow for NTSC! There's enough room on an NTSC carrier. We've demonstrated the ability to send a couple of hundred kilobits per second on an existing NTSC signal without disturbing any of the other components, including the MTS stereo [signal]. So you could have six-channel discrete digital sound on your TV! With a cheap chip to do the decoding, and a low bit rate, it [AC-3] can be piggybacked onto videodiscs, it can be back-engineered into VHS tapes, and things like that. So we are certainly looking at the possibility. This is, at the moment, a six-channel system. But our conversation has indicated that for any sort of "perfect" system you might want a couple of dozen channels. The techniques involved [with AC-3] are perfectly extendible. The same coder, in relatively few additional bits, could code 24 channels if desired. One of the things that digital coding offers that may not be immediately obvious to the casual observer is that the bit stream can be logically subdivided into any number of information streams. This is something that you can't do with analog. Up to now, all of the spatial audio systems that we have been messing with—stereo, quad—all try pretty much to piggyback the spatial information on the audio, such as matrixing the audio channels together or something like that. That of necessity compromises both the spatial information and the audio. An advantage of a digital channel is the ability to divide it very cleanly into separate information streams, sending the audio as part of one stream—and the spatial information, to the degree to which it can be segregated, can be sent as a separate data stream, where neither data stream corrupts the other. This is a very valuable and powerful solution. But to make use of it requires a substantial amount of work on optimum ways of coding spatial information into multiple data streams. Our AC-3 is our first effort along that line, and I can see the possibility of further and more powerful spatial coding as time goes on.

RANADA: Much of what we've been discussing revolves around the basic lower limit of the ear's data rate. What is the basic data rate of the ear? Even though you can code what was originally gigabits per second down to megabits per second without any perceptual difference, there must come a point at which you cannot code below—a minimum data rate.

DAVIS: One very simple way to estimate that, which will give a very conservative answer, is to say that the ear has approximately a 96-dB dynamic range (a "16-bit range"), and it has kind of a 20-kHz bandwidth. But if you just assume a channel can handle 16-bit data with 20-kHz bandwidth, that right there is 700,000 bits per second times two channels, giving 1.4 megabits per second—the CD data rate. If you allow for audio thresholds changing with frequency, you can knock off from that probably another factor of two and say 350 kilobits per second per each ear, total. But that's a fairly simplistic analysis, and I think it's conservative. *[David Ranada's Note: The point is that the entire infinite-microphone imaginary recording sphere talked about earlier should be codable down to this 350 kilobits/second data rate.]* Now I've recently heard claims—I think it was from Anderson from MIT—who claimed that in fact the internal data rate that you can actually absorb is way down at an unbelievable 100 bits per second! But someone pointed out that 100 bits means 100 flips of a coin every second. It's actually like you have 2-to-the-100th-power different possible outcomes, and you are differentiating one out of 2-to-the-100th outcomes every second. And you can't ask a person to do any more than that—it's hard to ask a person to do even that much. Two to the 100th power is a huge number. And yet, to do any of these coders, so far we still need many orders of magnitude more bits per second. Any time we try to get anywhere near even several thousand bits per second, much less several hundred bits per second, the audio quality goes away quite substantially, so far.

"I think [coding a very large number of related channels] is doable, and you'll see this coming along within the next five to ten years. And I'll be one of the people trying to make it happen."

RANADA: At 100 bits per second, the waveform entering your ear will not be anything like the original.

DAVIS: That's a question about what you do with those 100 bits. The thing is that to get down to substantially lower bits per second than we have now, we have to go to what might be called event coders. If someone strikes a piano note and it lasts 4.5 seconds and they let go of the key, you send that information. Here's a piano and it has this harmonic content and here's this note that lasted 4.5 seconds. And at the playback side you have this very elaborate decoder which amounts to a synthesizer that puts the signal back together. I think that this will be the sort of coders we will have.

RANADA: But this is less psychoacoustics and more synthesis...

DAVIS: But in order to do it properly the analysis of the original sound will have to be in psychoacoustic terms. You won't have the luxury of having an isolated piano note you can do this processing to. You're going to have arbitrary sounds, and the coder will have to break the complex sounds into parts that can be characterized psychoacoustically in very concise terms.

RANADA: And you think this is doable?

DAVIS: I think it is doable, and you'll see this coming along within the next five to ten years. And I'll be one of the people trying to make it happen.

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6. Interview with Bob Carver, Manufacturer and Audio Designer

RANADA: How did you get interested in audio?

CARVER: My mom was a musician and my dad was an engineer, so here I am. It sort of emerged naturally.

RANADA: Was there any revelatory experience that showed you the audio pathway?

CARVER: When I was about 13 or 14 years old, I heard a Heathkit hi-fi and it just bowled me over. I could not believe these incredible sounds that were coming out of this big giant speaker that was about the size of a refrigerator. I had never heard a high-fidelity system before and it was wonderful. I was hooked ever since then. I started reading magazines and stuff like that.

RANADA: But at that stage you were probably too young to know what you wanted to do with your life.

CARVER: No, actually, that's not true. I've known ever since I was little—since I was perhaps four—that my life was going to be having something to do with electricity. The way I expressed it then was that I was going to be an electrician. Electricians did interesting and fun things with electricity. So when I was a Cub Scout I got merit badges by building anything that had to do with electricity. As I recall, that included making a crystal radio set. It included making a capacitor out of tin foil, and other scientific experiments. So I knew I was going to be an "electrician."

RANADA: So you directed your higher education in that direction?

CARVER: Right.

RANADA: You have a degree in what?

CARVER: Physics.

RANADA: You specialized in electronics I assume?

CARVER: No, actually, the electronics was always an avocation. I've trained as a physicist, but circuit design is my first love.

RANADA: So you have a scientific training which gives you a slightly different background than is the case with those who have simply gone into engineering?

CARVER: I've found that to be absolutely true. Physicists learn a couple of things. One is not to take themselves very seriously. As soon as one tries to learn the secrets of the universe... I'll tell you something—studying the secrets of the universe is an extremely humbling experience. As we go along studying or learning the secrets of the universe, we learn another very important thing—that whoever designed this universe of ours really had his, or her, act together. We feel terribly insignificant just trying to understand it... Physicists learn to take themselves with a grain of salt and with a little bit of humor, hence we name things funny. Physicists always give things funny names, like quarks, charm, spin...

RANADA: Magnetic field amplifier...

CARVER: Oh yeah, magnetic field amplifier. That just goes to show that we like to have fun.

RANADA: How would you say you look at designing differently than a "normal" engineer?

CARVER: Well, I think I approach it from fundamental first principles. My design approach starts with first principles as opposed to perhaps a set of application notes. As a physicist, I have such a terrible memory, I never remember anything. I can't even remember the table of integrals, and that's terrible since it means that I always have to start—when I want to design something—from first principles. Like, for a capacitor, let's see, here's an electrical field between these two plates, and then I'd have to derive the equation on the spot because I can't remember it. I'm exaggerating a little bit but not much. So when I design something, it really does start from absolutely the ground up. And then as I go along and as I'm traversing the design stages, that's when I start looking up the work of others to see what they've done. But it's sort of halfway through the process.

RANADA: Let's go back to the very first Phase Linear product. How did that spring into your mind?

CARVER: As a hobby I had built amplifiers—tube amplifiers—and had even built a giant tube amplifier. So I knew about tube amplifier design. And when I decided I wanted to make a living designing amplifiers, I decided to do a transistor amplifier.

RANADA: What made you decide to make a living out of designing amplifiers?

CARVER: All through school, especially through graduate school, I found that designing amplifiers was so much fun. I can't describe how much fun it is to sit down at a workbench and design an amplifier and have it work. It's hard to describe. I have no idea why it is so much fun; it just is. The Phase Linear 700 was my first transistor amplifier. Again, starting from first principles, I didn't understand how difficult it would be to make it work, to make it hold together at the very high voltages that were required to produce a powerful transistor amplifier. My tube amp had 150 watts, but when I played it into a pair of loudspeakers and put a scope across the loudspeaker terminals, I found that the output voltage went very high. When I decided to make the transistor amplifier, I said, well, I'm going to make the transistor amplifier have as much voltage swing as my tube one. I found out quickly that to make that voltage swing it had to be 350 watts into 8 ohms. There were a couple of things that made the Phase Linear possible: the high-voltage transistors that Delco made for automobile ignitions and a comprehensive protection scheme that made the thing "flameproof." And that protection scheme has subsequently been copied by almost every amplifier manufacturer there is, at least until recently. I call it an energy limiter. It turns out that the thermal time constant of a transistor chip, in a TO-3 transistor package, is about the same time as one quarter of a cycle of 20 Hz. And if you play music, it turns out that all you have to do is design a protection circuit that will allow that large volt-time integral to pass through the transistor chip. If you make it be approximately the same as the thermal time constant, it'll work. You can play music and the amplifier won't have to go into limiting. Prior to that time, transistor amplifiers either had to have a loophole in their protection circuits whereby an unwanted protection initiation would occur and cause a snapping sound in the music, or a loophole had to exist in the protection scheme itself whereby, if you shorted out the amplifier or hooked it up to a faulty load, the amplifier could blow up because it was inadequately protected. In fact, the Crown DC-300 used to have a switch on it labeled "hysteresis" and "normal," and the instruction manual that came with this beautiful Crown DC-300 amplifier said, when you're first hooking up the amplifier throw the switch to "normal." That gives you the maximum protection. If you accidentally short it out while you are hooking it up and first testing it, you won't blow it up. Once you are secure in the installation, you are supposed to throw the switch over to hysteresis so you can drive a load. But don't tempt fate. The fact that the switch even existed indicates that protecting an amplifier back in those days was a real problem. So that's why I came up with the energy limiter. It's really an obvious concept, but my Phase Linear happened to be the first one incorporating it.

"I can't describe how much fun it is to sit down at a workbench and design an amplifier and have it work. It's hard to describe. I have no idea why it is so much fun; it just is."

RANADA: The next thing was a preamp with many innovative features in it. How did that come about?

CARVER: That harks back to my tube amp days when I had my hi-fi set in my small workshop. And I would listen to my Beethoven symphonies with Bruno Walter conducting, and there was a lot of hiss. There was so much hiss it was unbelievable. I thought, boy, somehow I've got to figure out how to make this hiss go away. One day I sat there with my hand on the treble control and I noticed I could turn the treble up when the music was loud and I could turn the treble down when the passages were low. As I listened to the symphonies a lot I got to the point where I would sit there with my hand on the treble control and gain-ride the treble. When the music played loud I'd turn the treble up so that the highs were there. And when the music was soft I turn it down so the hiss would go away. I thought, "Aha! This is really slick." From that developed the Autocorrelator, which, as you know, is a series of voltage-controlled bandpass-filter gain stages that are sensitive to the musical spectra. Actually it works kind of like the way the new DCC works.

RANADA: I remember your getting a little flack for the name Autocorrelator. You often get into trouble with the names you pick for things.

CARVER: Physicists always pick good names for things. Autocorrelator seemed like a good name to me. Actually, there is a basis in fact for the name because one of the problems of controlling a series of a bandpass gates is to dynamically raise and lower the thresholds at which they operate. And those thresholds are related, from microsecond to microsecond, to our hearing thresholds, which change with the musical spectrum because of masking. I'll use two extreme cases to illustrate this point. If you play white noise, there's no point in having a noise-reduction system. But if the musical information you are listening to is, say, a struck glass goblet ringing with a nice pure tone, more or less, it's going to be easy to hear any hiss surrounding that tone. It turns out that when a signal is highly correlated, like a turning fork, it's necessary to have the gate thresholds high, otherwise you'll hear the hiss. But when your are listening to music, the correlation coefficient of the music is changing from moment to moment. As an example, consider human speech. During the sibilant portions of speech, the correlation coefficient is extremely low—approaching zero—so the gate thresholds under those conditions can go down. Let's say you pronounce the words *speaking softly*—during the *s* sounds the thresholds can be very low because the masking is very strong, but during the other portions of the phrase the thresholds have to go up. The trick was to control the gate thresholds dynamically along with the information in the music or voices. There's a circuit that determines the correlation coefficient of the incoming signal. That circuit in turn becomes a control voltage that sets the gate thresholds. If the sound is highly correlated, the thresholds go one way and vice versa. So that's why I call it an autocorrelator. The inventor gets to call it anything he wants.

RANADA: There's also the very famous Sonic Holography circuit. How did you decide to do something like that? There were papers available on similar signal-processing techniques. Did you read them or did you start with first principles again?

CARVER: I went again from first principles. After Phase Linear, when I wanted to make a comeback with Carver [Corporation], I figured I needed a couple of new technologies to have people pay attention to me. At first I was thinking of just one, but then I thought two would be better for good measure. One was the Mag amp and the other was Sonic Holography. Over the years I had become less and less enchanted... I knew there was something wrong with stereo. I understood early on that stereo presents us with only a limited set of cues and clues that prevent our ear/brain system from developing a believable sense of a acoustic space. That's because, when this stereophonic system that we enjoy was conceived, very little thought, if any, was given to preserving timing cues. All of the thought was given to preserving amplitude cues, that is, left/right separation in the stereo system. Today, left/right separation is no big deal in any stereo system. But achieving a sense of spaciousness and depth is a big deal and, in fact, high-end audio in many respects has grown up in the search for that difficult-to-find Holy Grail—that is, a sense of depth and spaciousness. You can read in high-end audio journals that "this amplifier gave a sense of layered depth; I could tell which violin was in front of which violin." So a sense of depth is obviously searched for, and people die for it and so on. It's hard to get and the reason is that our stereo system was designed without providing the proper set of spatial cues and clues, the so-called timing cues. For our brain to latch onto a set of timing cues, it's necessary that each ear receive the correct temporal information, which is not possible with a simple pair of speakers. For example, in real life, when we hear a sound it has the proper amplitude cues and the proper temporal cues for each sonic event. We hear sound arrivals, and the temporal displacement of those arrivals gives us the timing cues and allows us to locate the sound in three-dimensional space. But when that sound is played back over a set of loudspeakers, instead of a pair of sound arrivals (one for each ear from a single sonic event), we have four (two per speaker per ear). Four arrivals into our ears instead of two is incorrect and our brain doesn't quite know what to do with it. Our brain has evolved through evolutionary millennia to work with two arrivals, not four. So what happens is our imagination has to work overtime to make us believe that a stereo representation is real. A child can listen to it and say, "Aha! That's not the real thing. I'm not fooled!"—regardless of how good the equipment is. So, Sonic Holography was a solution to the extra arrivals. It simply acoustically canceled the unwanted arrivals so that we're left with two. And two arrivals are much more believable than four arrivals. And the expression Sonic Holography of course derives from a hologram, which builds a virtual image where none really exists. An optical hologram does this by combining two beams of light in an interference pattern, a reference beam and another beam, and by constructive and destructive interference an image is built. Sonic Holography works sort of the same way. Two independent sounds—one from the left speaker and one from the right speaker—are combined in space around our head. Constructive and destructive interference cause the correct-side sound to be accentuated and the incorrect-side sound to be canceled. So the term Sonic Holography refers to two things. First, people will hear the expression Sonic Holography and go: "I know what that's about; it's about images." And, of course, sonic means that this is for sound. And it does work in a similar fashion to a visual hologram.

[Bob Carver's carefully constructed rationale for Sonic Holography is somewhat vulnerable to an argument I have been throwing at him for years, namely that the producer and engineer of a typical stereo recording hang their microphones and create their final mix to make the sound as believable as possible through the standard two loudspeakers producing four arrivals. Removing two of those anticipated arrivals alters the fine-tuned compromise and may under certain circumstances be counterproductive. With some recordings it works great.—Ed.]

RANADA: These innovative circuits were all designed to solve very specific problems which existed ten years ago. The audio scene is now very, very different. What do you see are the problems remaining, or where can your inventiveness be applied now?

"...when this stereophonic system that we enjoy was conceived, very little thought ... was given to preserving timing cues. All of the thought was given to preserving amplitude cues..."

CARVER: My big thing is psychoacoustics. And psychoacoustics is an area that audio designers have really not paid much attention to, in my opinion. I believe that is because it is a very difficult arena to play in. We have to understand how we hear things, why we hear things, and know how to manipulate that without fear of doing something wrong in the process.

RANADA: What is wrong, if what the end result sounds like is what counts?

CARVER: That's my belief. There is nothing wrong with that, of course. But you have to be fearless to even have that notion, it turns out, in today's audio world.

RANADA: Why would you have to be fearless if you're working scientifically?

CARVER: There's a notion that what appears in the record groove or the CD bit stream is somehow pristine and perfect. That if only the amplifier and the preamplifier and the loudspeakers could preserve this pristineness, then we would have believability in the sound field. That somehow the sound coming off the disc is so perfect and oh-so-delicate, and if you mess with it you're going to screw it up and ruin the believability. This is a belief system that seems to have gained momentum in the last decade.

RANADA: Is this due to the influence of high-end magazines?

CARVER: I believe it is. I believe that people just entering the fascinating field of audio are hungry for information and that the notion of a pristine signal is severe misinformation. It turns out that the information coming off the disc is not all that pristine. It's quite flawed to begin with and, further, it's not delicate. It's not going to be damaged by looking at it the wrong way. You have to be fearless in that you understand that the information coming off the disc is flawed to begin with, at least in format, and not be afraid to work with it, to investigate it, to understand it, and to change it so that when it is played back through a stereo system it'll produce a more believable image of reality.

RANADA: So you are willing to forgo what would normally be called accuracy in order to obtain realism?

CARVER: I think accuracy and realism go hand in hand. You can't have realism without accuracy but you can certainly have a lot of accuracy without any realism. So the comparison is an apples-and-oranges comparison. But back to the original question. The audio scene has changed a lot. We have super-power amplifiers, we have loudspeakers that are better than ever, everything's better than ever. You might say, well, what's left?—like the proposal to shut down the patent office many years ago because everything had already been invented. It turns out, I think, that the things that should be tackled today are the psychoacoustic events that lead us into believing that a reproduced soundstage is real. And there's a lot of work to be done in that area. I believe that it's going to involve somehow the superposition of acoustic vectors in space for the listener so that the sound vectors will be essentially the same as they were in a real-life recording venue. But there's an important distinction here. For something to be seen as if it were real it doesn't have to be facsimile. It really doesn't have to be identical. But the way our brain reads it, it has to sense that it could have been real. And that's the important thing to remember. You know if you go to the Holodeck—the computer-constructed illusion-recreation area of the U.S.S. Enterprise in *Star Trek, the Next Generation*—the Holodeck illusion seems very real even though it never existed in reality, and it's real to the touch, the smell, the feel and all of that. And that's what I'd like to do with the musical experience in our listening room—basically, have a musical Holodeck. It may not be a facsimile reproduction of anything that actually existed but it's so real to the senses that it certainly could have been.

RANADA: In a lot of this kind of effort, the problems are so enormous that it would require a massive R & D effort, mountains of DSP and computer power to do this kind of stuff. Are you in a position to do this?

CARVER: Yeah. Powerful computers are cheap now. And the beauty of audio design is that it doesn't take an intensive capital investment. It's not like building an airplane where you have to have all this machinery. You have a desk to work on, a computer, somebody to help you assemble circuits, a soldering iron, a scope, and a few other odds and ends. It's basically very cheap.

RANADA: So you don't feel intimidated by the immense resources some companies can bring to a particular problem?

CARVER: No, I don't feel intimidated; I feel jealous. Obviously, developing the CD system is beyond my scope, but that's also not my specialty. And I tell you the wonderful work that Philips and Sony have done on the DCC and MD is so refreshing—to see somebody tackle it and do some tremendously good work in that area.

RANADA: You mentioned high-end magazines a while back. Have high-end magazines been good or bad for the industry?

CARVER: I think that they've been both good and bad. On balance, though, I think they've been more good than bad. They've been good in developing a language of high fidelity. For example, Harry Pearson of *The Absolute Sound* was the founding father of the language that audiophiles use to describe their experiences; in many ways he genuinely was. And, even before him, Gordon Holt I remember one time published a little dictionary of audio terms, and he used expressions like *woolly* for the bass, and *fuzzy.* And then Harry Pearson elevated that and was a real class act in developing a language for high-fidelity that was understood throughout the world. As a single man he was very, very responsible for giving us a language to talk about high fidelity. And also, he and others like him taught how to appreciate the dimensional aspects of a soundstage. In fact, he wrote an article on the stereo soundstage which had influenced me when I was just getting into appreciating this. So the high-end audio journals taught us how to listen; they've taught us what to listen for; they've been so entertaining as not to be boring. So we've stuck with it and we've learned. Where they've been an abominable failure is in the scientific disinformation and misinformation, and that's been terrible. Some of these crazy notions that are absolutely off-the-wall with no scientific basis.

RANADA: Do you think these notions are dangerous?

CARVER: I think that whenever fantasy masquerades as truth it's dangerous. And especially when it becomes a part of a belief system, because people are willing to die for belief systems no matter how right or how wrong. What happens is that people who are entering audio for the first time really want to find out about hi-fi. And they read anything and everything they can get their hands on. When somebody says you should freeze a CD, or you should plug this special magic clock into the wall so your system will sound better, they can't tell that that's not true. They believe it. These are very smart people and very bright people, and they want to have an open mind. So they don't close their minds to these notions. But it's really an emperor's-new-clothes kind of thing. I think there's some damage done. But on balance I think it's good.

RANADA: Do you feel that you've been done wrong by some high-end magazines? Have you been damaged by them?

CARVER: Oh absolutely, tremendously damaged. Because when a high-end magazine says that the Carver amplifier, because it's so small, must not be any good, and then listens to it and says this thing sounds terrible—even though it has more current, has more voltage, has more power, can drive lower-impedance loads than anything else around, for a fifth of the price, and can even have tube output characteristics so it can sound like tube amps—and make pronouncements like that when they're patently incorrect, that's harmful because many people believe it. It's like a restaurant reviewer turning up his nose at an inexpensive dish even though it might be a great, great dish. It's not fair, but that's life.

"I think that whenever fantasy masquerades as truth it's dangerous. And especially when it becomes a part of a belief system, because people are willing to die for belief systems..."

RANADA: You learn to live with it?

CARVER: No, I haven't learned to live with it. It hurts my feelings. But what can I do? Actually, there's a lot I can do. I can educate people. People want to understand how the world works, and if somebody out there tells them how the world works they'll latch on to it, make sense out of it. I think that there's a lot that can be done and that I can do here.

RANADA: Where do you see the audio industry going?

CARVER: I think what's happened is the ratio of people interested in general sound compared to component audio has gone up. In other words, as a percentage of the total population interested in audio, a smaller percentage is interested in component audio. But the total population has grown, so the interest in component audio has grown. You can sort of verify that. Take a look at the proliferation of high-end audio companies that are in the one- to four-million dollar range in sales. There are lots of companies like that making high-end audio components, and they are enjoying success. Let's go back to the mid-to-early '70s; hardly any of them existed. There was a much, much smaller number of them. So it's grown. It really has. Just in my block, I sometimes during the summer leave my doors open and play my stereo. Like in the Pied Piper story, kids come in from the neighborhood going "Oo, ah, wow!" They want to know all about it; they get sucked in.

RANADA: Do you think audio will ever reach a peak of development after which further progress would not be possible?

CARVER: A deep, deep question, the answer to which would be very presumptuous, almost dangerous. But... if it's possible to develop a stereo system in which one could close one's eyes, sit back and not be able to distinguish whether you are actually in the presence of a real-live musical performance or in the presence of a sound system—if it's possible to attain that, then you have to say, "Well, that's it, you don't have to do any more work." The problem is, imagine in your mind's eye that the hardware for such a system is successfully developed. There's then the problem with the software; you'll bring home records and some of them will sound real and some not so real. There will always be more work to be done. When it doesn't sound so real, there'll be the temptation to tweak one's system. So the fun will always be there, and I don't foresee a solution in my lifetime.

RANADA: Do you think, on an abstract level, it is a solvable or unsolvable problem?

CARVER: I think it's solvable.

RANADA: Some people say that humans will always be better than the equipment they are listening to. But one can take the opposite viewpoint and say that the equipment is already better than what humans can do, and the problem is what's being fed through it and the exact way the circuitry is employed.

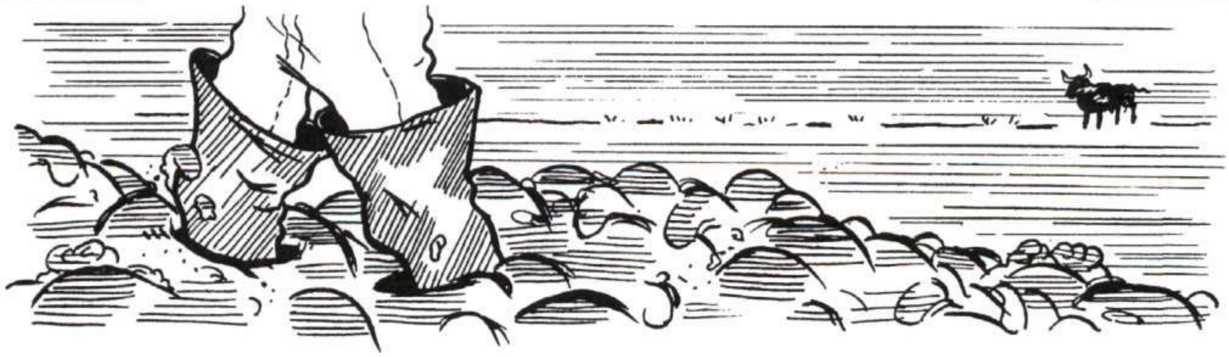
CARVER: That's exactly right. The concept of limits has to be addressed. The notion that humans are always better than the equipment ignores the limits that are associated with human beings. And there are limits. There are limits to all of our senses, and those limits are very well understood—or at least sort of well understood—and certainly well defined. So we'll know when the equipment is better than it has to be; we'll know when the equipment is better than the limits that our senses are able to deal with. Again the notion of limits is an important one because it is given zero—literally zero!—credence by many high-end audio journals.

RANADA: Do you think the equipment will be good enough in your lifetime?

CARVER: Yes, I'm hoping. I want to be part of making it good enough in my lifetime; that's my plan. •

Hip Boots

Wading through the Mire of Misinformation in the Audio Press



***Editor's Note:** I turned my pet column here over to David Rich for this issue because he is so upset about the monstrously ignorant "technical" articles currently appearing in various high-end audio publications that he needs a printed outlet for his pent-up indignation on the subject. As he pointed out to me when he requested this space, we aren't dealing here with the slower students in an engineering class but with self-appointed "experts" who take your money for their stupidities. I agree. I don't think all of our readers are fully aware of the utter contempt of degreed engineers and responsible audio professionals for the untutored scribblings of these pitiful pundits. "How do they get away with printing such garbage ? " those credentialled authorities keep asking me. How indeed.*

Gerard Rejskind in *The Absolute Sound.*

The depths to which the underground journals will sink to find someone to support the claim that linear PCM coding is fundamentally flawed seem to be bottomless. In choosing Gerard Rejskind, the editor of *Ultra High Fidelity,* a Canadian underground journal, to address this topic, *TAS* has sunk lower than ever before. [Rejskind, G. "The Sound of Digital: The Present State-of-the-Art." *The Absolute Sound* 17.81 (July/August 1992): 28-36.] Mr. Rejskind's qualifications as stated in the article are: "I have... listened to countless [CD] players...and I have had the privilege of talking with numerous designers." The resulting *TAS* article is as flawed as any article I have encountered that claims to be a scientific analysis. Not even Bob Harley has so far written anything quite as absurd.

Mr. Rejskind states: "The usual claim of over 90 dB of dynamic range is based on a common mathematical blunder, coupled with what may be outright fraud." After this statement he goes on to present the wrong formula for the signal-to-noise ratio (dynamic range) of a PCM system, namely 20 log $(2^b - 1)$, instead of the correct formula, which is $6.02b + 1.76$. He then goes on to state the formula is in error because the noise level is calculated as a peak, not rms, value. Apparently he never looked at the derivation of this formula in any standard text on communication systems or data conversion, since such a text would have shown him that this statement is completely false, in addition to showing him he was using the wrong equation. Mr. Rejskind goes on to make the completely false claim that the last bit of the 16-bit word in CD coding is a parity check bit and only 15 bits of the word are data. Mr. Rejskind blames the overstatement of signal-to-noise measurements (such as "over 90 dB") in CD players to the use of a zero-code digital data stream to make the measurements. The fact that sine wave tracks are used to assess the signal-to-noise plus distortion characteristics of a CD player is known to any reader of any audio magazine that performs electrical tests on CD players. Such a reader also knows Mr. Rejskind's statement that "only the best players can reproduce a sine wave at a level of -60 dB as anything but a caricature of the original" is also completely false. But to quote Mr. Rejskind, "don't put away your calculator just yet, because the fun is just beginning."

Gerard Rejskind uses the techniques of a patent medicine salesman to claim that inband harmonics occur when an unquantized sine wave is sampled above the Nyquist rate. He offers proof of this by invoking concepts for the analysis of amplitude modulation waveforms and applying that approach to the points of a 12.5 kHz sine wave sampled at a 44.1 kHz rate. He then claims to identify a sideband at 2.321 kHz. It all looks very correct to those unaware of the sleight of hand Mr. Rejskind is using. The fact is that amplitude modulation theory simply does not apply in this case. Interested readers looking for a correct explanation, at a layman's level, of the sampling theorem (Mr. Rejskind spells it *theorum)* are directed to the Ken Pohlmann text, *Principles of Digital Audio.* Another example of this sleight of hand can be seen in the following excerpt from Mr. Rejskind's article: "Sophisticated designers...know that no [digital error] correction system can be trusted to fix gross errors. Sensible amplifier designers are aware that they must build circuits that work well even without feedback..." Mr. Rejskind hopes the reader is unaware of the fact that analog feedback theory and digital error correction are totally unrelated fields.

Figure 1 of the article shows that Mr. Rejskind's ego has no bounds. The first part of the figure shows the analog and digital sections of a CD player connected through zero-impedance ground connections. The caption of the figure is, "How designers think they ground their circuits." The second part of the figure adds resistors to the ground leads. The caption reads, "How they actually ground them." Come on, Mr. Rejskind, any C-minus student in the second year of E.E. school knows that ground connections are not necessarily zero-impedance and may cause problems. According to Mr. Rejskind you can hear the effect of the ground interaction by slapping the CD player. You can then, according to him, hear jitter caused by this interaction. The fact that the player's sound changes because the error interpolation circuitry is activated and the disc synchronization signals are momentarily lost when the CD player mechanism is jarred apparently never occurs to him.

From the above it should be clear that it is Mr. Rejskind who is in effect committing "what may be outright fraud."

—David Rich

Robert Harley in *Stereophile.*

You may ask why we keep coming back to Bob Harley in this column. The answer is that he has become the most widely read and followed journalist in the field of digital audio. According to manufacturers, a bad review by Harley, no matter how inaccurate, will send the sales of the product to almost zero. That makes it impossible to ignore the dramatic errors in his copy that occur because of his lack of training in electrical engineering and his desire to believe in every claim for something that makes an audible difference.

As an example, he virtually paraphrased the press release for Sony's Super Bit Mapping (SBM) noise shaping system. ["Industry Update, US: Robert Harley." *Stereophile* 15.8 (August 1992): 53-57.] He even included misleading figures supplied by Sony. He described the system as a Second Coming, with dramatic improvement in sound quality. Any competent audio engineer reading the Sony press release would have realized that the concept of noise shaping was not new and that the significance of noise shaping was being seriously misrepresented, wittingly or unwittingly, by the author of the press release. So inaccurate was Harley's original article that he was forced to retract the claims he had made there after Dr. Stanley Lipshitz of the University of Waterloo and Robert Adams of Analog Devices had explained to him his errors. ["Industry Update, US: Robert Harley." *Stereophile* 16.1 (January 1993): 51-55.] Apparently Harley felt no need to talk to these experts before running the original article because he thought he heard a dramatic improvement in the sound quality, and this alone validated any claims by Sony. My question is, now that Bob Harley knows that the SBM system is much less than it first appeared to be, does he still hear such dramatic differences in the sound?

In another article, Harley attempts to explain the design innovations made by Robert Gendron in the new Museatex Audio digital processor. ["Industry Update, Canada: Robert Harley." *Stereophile* 15.9 (September 1992): 47-51.] Here it not a case of Harley parroting a press release; instead, he fails to explain the operation of the system because it is clear that he has not understood what was being explained to him. In this article Harley shows that he has no idea what the difference is between an finite impulse response (FIR) digital filter and an infinite impulse response (IIR) filter. He then goes on to claim that the Museatex oversampling filter switches between both FIR and IIR filters, when in fact it uses only FIR filters in a novel algorithmic adaption scheme and does not use an IIR filter at all. His explanation of Mr. Gendron's new S/PDIF decoder is so garbled as to render the idea virtually unrecognizable.

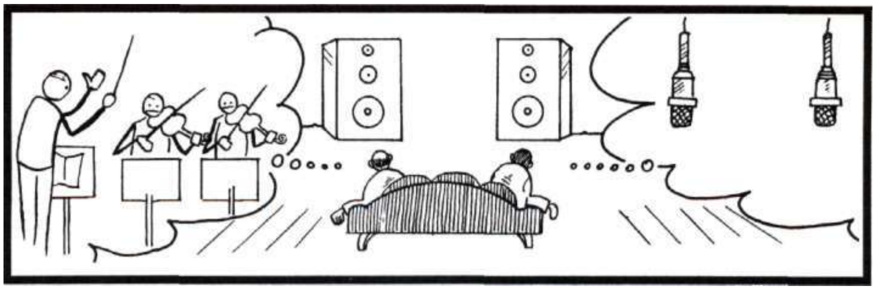
I obtained an explanation from Museatex of how this novel S/PDIF decoder works, but since some of the technical material that was discussed with me may be proprietary I will not comment on the decoder here. I will give more details on these design innovations if Museatex decides to send us a sample of the new D/A converter. The company has already supplied one to Bob Harley, apparently because his power to make or break a company outweighs his manifest inability to explain what the company is doing. That is really a shame because the ideas embodied in this product appear to be truly innovative and important, and deserve to be presented lucidly.

—David Rich

[I wonder how Larry Archibald, Stereophile*'s owner and President, is able to look himself in the mirror in the morning when he is shaving and tell himself that he is running a credible and responsible publication. He is being told over and over again—not just by us but by strictly neutral and disinterested parties with impeccable technical credentials—that Bob Harley is a loose cannon in his organization and yet he does nothing about it. Not that Harley is the only one at* Stereophile *writing technobabble, but in his case the management appears to regard that as part of his official job description.*

—Ed.]

Recorded Music



The plan for this column was to have David Ranada and others take it over completely, but David got busy with higher-priority assignments (although he keeps threatening to do a compleat Stravinsky Le Sacre du Printemps *disco graphy for us here), and those others—where are you? So, it's your overburdened Editor "once more unto the breach."*

Catching Up on the New and Not-So-New CDs

By Peter Aczel
Editor and Publisher

The condensed tabular review format I experimented with in Issue No. 17 attracted favorable comment from all those who commented at all; it seems that broad horizontal coverage is more in tune with the needs of our readers than in-depth reviews of fewer releases. After a large accumulation of new CDs on my shelves, I decided to go back to capsule reviews; however, I had found the tabular format with its precisely defined fields (in the data-base sense) to be too rigid as well as needlessly repetitious, so a capsulized version of my earlier columns seemed to be worth trying. I went back to the label-by-label approach (as against composer by composer) because the comments on producers, recording engineers, recording techniques, etc., are generally applicable to all recent releases under the same label and need to be made only once; it's a more efficient and concise way of organizing the reviews. Besides, audiophiles are very label-conscious.

I no longer list the SPARS code for each CD because it's nearly always DDD; only the exceptions need to be noted. The year in parentheses after each listing always refers to the recording session or sessions, not the release or copyright date.

The 20-bit bandwagon.

The hot button in the CD world right now is the conversion of 20-bit digital master tapes, now coming into wide use as the professional studio standard, to the required 16-bit standard used in CDs. It looks as if every company were getting into the act and making sweeping claims about its special proprietary technology for doing this, whereas in reality all of them are obviously using very similar though perhaps not identical noise-shaping techniques. (See also the "Hip Boots" column in this issue.) So far I have heard the good news from Sony (formerly CBS), Reference Recordings, Telarc, and Dorian, but I know that there are others. I don't think it's as big a deal as they want the world to think but it's certainly a desirable development because it makes the achievement of genuine 16-bit resolution in the final product easier and therefore more likely. I seriously doubt, however, that one can hear the difference between perfect 16-bit and perfect 20-bit encoding/decoding of recorded music, but maybe one can between imperfect 16-bit and careful 20-to-16-bit. We shall see; so far there has been only a very thin trickle of the new CDs and nothing of major importance.

Bainbridge

The following two audiophile spectaculars are actually the work of Mantovani Production Associates, recorded in London with different production and engineering teams, and distributed by Bainbridge.

"The Age of Swing," Volume 1. The BBC Big Band. BCD 6291 (1991).

An authentic and stylish re-creation of the big-band sound of the '30s and '40s by the almost too virtuosic BBC players. The sound is close to perfection—up-front, very clean, and very dynamic.

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"Golden Cinema Classics," Volume 1: The Adventure Film. The BBC Concert Orchestra. BCD 2521 (1992).

It opens with a "James Bond Suite" and ends with the music from *Ben Hur.* If you like that sort of thing, you'll love this CD. The orchestral playing is first-rate, and the sound is again super, in an appropriately unsubtle way—big soundstage, high definition, in-your-face brass.

Delos

John Eargle is still the chief recording engineer at Delos, and in my book he is still *numero uno* in the classical music field. Others may on rare occasions equal him or even surpass him, but they're not nearly as consistent. The company, under the leadership of Amelia Haygood, is further distinguished by its unique efforts on behalf of mid-20th-century American music, long neglected.

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Paul Creston: *Symphony No. 3, Op. 48; Partita for Flute, Violin & Strings, Op. 12; Out of the Cradle; Invocation and Dance, Op. 58. Seattle Symphony, Gerard Schwarz, conductor. DE 3114 (1991-92).*

Paul Creston used to be performed almost as frequently as Aaron Copland; then his conservative but robust, colorful, and beautifully crafted music—admittedly not as exciting as Copland's—fell out of favor. These are meticulous performances, and the sound is as panoramic, open, and finely detailed as you'd expect from John Eargle.

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Howard Hanson: *Symphony No. 7; Symphony No. 5, Op. 43; Piano Concerto in G Major, Op. 36; Mosaics [1958]. Seattle Symphony & Chorale, Gerard Schwarz, conductor; Carol Rosenberger, piano. DE 3130 (1992).*

This is mostly late, or late middle, Hanson and not quite as unequivocally convincing as some of his earlier work. The *Mosaics* variations are quite powerful, though, and Carol Rosenberger in the concerto is sensitive and surehanded, as always. The sound quality throughout is strictly late Eargle, meaning high-end demo quality.

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Walter Piston: *Symphony No. 4; Capriccio for Harp and String Orchestra; Serenatafor Orchestra; Three New England Sketches. Seattle Symphony, New York Chamber Symphony (Serenata), Gerard Schwarz., conductor. DE 3106 (1990-91).*

Some of the earlier releases in this Piston cycle on Delos presented more exciting music, but completeness is the goal here, and Schwarz/Seattle are as good in this sort of thing as anyone could ask for. The concluding *maestoso* movement of the New England piece is alone worth the price of admission, and John Eargle's recording is once again glorious.

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William Schuman: *Variations on "America" (Ives, 1891; orch. by Schuman, 1963); New England Triptych; Symphony No. 5; Judith (Choreographic Poem for Orchestra). Seattle Symphony, Gerard Schwarz, conductor. DE3115 (1990-92).*

Stiff structures and superbly crafted sound, without much feeling—that's my image of Schuman (the one with only one *n*), but what a vehicle for a John Eargle audiophile spectacular! The orchestrated Charles Ives organ piece is loads of polytonal fun, though (hey, guys, the *tune* is "God Save the King," not "America"—how provincial can you get?), and the orchestral climaxes throughout the CD are breathtaking.

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Richard Strauss: *Macbeth, Op. 23; Serenade in E-flat Major, Op. 7; Ein Heldenleben, Op. 40. Seattle Symphony, Gerard Schwarz, conductor. DE 3094 (1990).* ***Richard Strauss:*** *Der Rosenkavalier, Op. 59, 1st & 2nd Sequence of Waltzes; Burleske for Piano and Orchestra; Die Frau ohne Schatten, Op. 65, Symphonic Fantasy. Seattle Symphony, Gerard Schwarz, conductor; Carol Rosenberger, piano. DE 3109 (1991).*

Gerard Schwarz's franchise in the standard Austro-German repertoire is not as strong—at least in the opinion of many discriminating music lovers—as in contemporary American music, but here he and the Seattle forces provide a pleasant surprise. *Heldenleben,* for example, is played with precision and carefully gauged expressiveness but without exaggeration—a fine effort, and so are the other pieces. In terms of sound, you won't find better Richard Strauss CDs than these two, although the composer's massed violins are more difficult to record with accuracy and beauty than the sparer textures of Piston or Schuman—but John Eargle knows how.

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Igor Stravinsky: *Le Sacre du Printemps (The Rite of Spring); Pulcinella (complete ballet). Seattle Symphony, Gerard Schwarz, conductor, with soloists. DE3100 (1990).*

The performance of the *Sacre* is a bit too careful, though precise and finely balanced (maybe only the absolute best orchestras can play it savagely *and* precisely). The *Pulcinella* is not only complete but very faithful to the 18th-century spirit and inspiration of the work. John Eargle's recording of the *Sacre* is unusually spacious and at the same time has explosive "bite," not an easy thing to achieve; the *Pulcinella* was done without him by his associates and very nicely, too.

"American Diva." Arias by Verdi, Puccini, Cilea, Charpentier, Wagner, Catalani. Alessandra Marc, soprano; New Zealand Symphony Orchestra, Heinz Wallberg, conductor. DE 3108 (1991).

What a surprise! What a voice! Alessandra Marc may not be a truly great artist—not yet, anyway—but in terms of voice production, power, range, and just sheer beauty of tone she is the equal of anyone, anywhere. Her top notes are sensational. The orchestral accompaniments are a bit on the plodding side, and the sound—done without John Eargle, in Wellington, New Zealand—is okay but not great, better for the voice (fortunately) than for the orchestra. Regardless of that, if you're an opera lover, get this CD!

"L.A. Guitar Quartet: Dances from Renaissance to Nutcracker." The Los Angeles Guitar Quartet (John Dearman, William Kanengiser, Scott Tennant, Andrew York). DE 3132 (1992).

Frankly, I didn't expect this to be as delicious as it is. Tchaikovsky, Praetorius, Gabrieli on four acoustic guitars? Well, it's a blast! These four musicians play with such fantastic rhythm, phrasing, and color that the experience is not to be missed. Don't be an uptight purist; give it a listen. Absolutely superb sound, too, by John Eargle.

Denon

This is the label of Nippon Columbia, but unlike other major international record companies they don't have a "corporate" sound dedicated to marketable mediocrity. Some of their best recordings sound as good as the proudest products of the small audiophile labels—and, of course, the artists are generally of greater stature. Denon's recording team appears to vary from session to session, but the name most frequently listed in the credits is that of Yoshiharu Kawaguchi as producer. Brüel & Kjær microphones are standard in Denon recordings, and normalization to a single microphone location by digital delay is still the technique used, to the best of my knowledge. Denon has never given up high-frequency pre-emphasis on their CDs, and I'm all for it (as long as the de-emphasis in the playback is accurate).

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Béla Bartók: *Music for Strings, Percussion & Celesta, Sz. 106; Concerto for Orchestra, Sz. 116. Orchestre de la Suisse Romande, Eliahu Inbal, conductor. 81757 9044 2 (1989 & 1991).*

Two of the towering masterpieces of twentieth-century music, both in slower performances than the composer specified. The MSP&C, however, is so finely detailed in texture and so lovingly phrased that I can't resist it, especially since the recording is wonderfully transparent and silky, with a very strong bass line. The Concerto is the earlier taping of two and could use greater virtuosity in the orchestral playing; it is also less impressively recorded. Even so, a recommended CD.

Joseph Haydn: *Symphony No. 6 in D Major ("Le Matin"); Symphony No. 7 in C Major ("Le Midi"); Symphony No. 8 in G Major ("Le Soir"). Orchestre de Chambre de Lausanne, Jesús Lopez-Cobos, conductor. 81757 9612 2 (1991).*

Early Haydn is different from the late symphonies everyone knows—more concerto-grosso-like, more coloristic, with lots of virtuoso solo passages. These are very well-played performances, thoroughly musical, not at all hard-driven, authentic in style and sonority albeit with modern instruments, and very pleasant in sound.

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W. A. Mozart: *Symphony No. 23 in D Major, K. 181; Symphony No. 28 in C Major, K. 200; Symphony No. 35 in D Major, K. 385 ("Haffner"). Sinfonia Varsovia, Emmanuel Krivine, conductor. 81757 9884 2 (1990).*

More of the same as the Krivine/Warsaw CD reviewed in Issue No. 17, but who can have enough of this when it's so nicely done? I find Krivine's approach to Mozart vigorous and at the same time sensitive. The "Haffner" is the highlight here. Again, excellent sound—suave and transparent.

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Arnold Schönberg: *Gurrelieder. Frankfurt Radio Symphony Orchestra with solo singers and 3 choruses, Eliahu Inbal, conductor. 2 CDs: 81757 9066 2 (1990).*

This supercolossal, oratorio-like, late-romantic, quasi-decadent work from 1913 (I'll see your Wagner and raise you a Strauss and a Mahler) needs a very firm hand and an illuminative musical mind on the podium. Inbal has both. If his singers were better, this would be a great performance; even as it is, I found it very rewarding. The sound is wonderful in the quieter passages, but the big climaxes are just a little bit strained.

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Dmitri Shostakovich: *Symphony No. 5, Op. 47. Frankfurt Radio Symphony Orchestra, Eliahu Inbal, conductor. 81757 4175 2 (1988).*

For some reason I didn't review this extremely lucid, lovingly shaped, perhaps too careful performance of one of my favorite Shostakovich works when I received it two years ago. The sound is very similar to that of Inbal's outstanding Berlioz *Symphonie Fantastique* of the previous year, as good as a Denon can get, but the Berlioz is more tautly, more excitingly played. The music itself is Stalin's only beneficial legacy—his brutal, ignorant meddling was the indirect cause of its composition.

***Karol Szymanowski:** String Quartet No. 1 in C Major, Op. 37; String Quartet No. 2, Op. 56. Carmina Quartet: Matthias Enderle and Susanne Frank, violins; Wendy Champney, viola; Stephan Goerner, cello. 81757 9462 2(1991).*

Modernized yet still overripe romanticism combined with an occasional touch of almost Bartókian acerbity but without Bartók's fascinating rhythms, surprises, and bigness of concept—this is not my favorite music but I can see why some like it. An early Webern quartet movement is also included as a filler. The playing is extremely polished, and the string sound is wonderfully gutty and vivid.

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***Peter Ilyich Tchaikovsky:** Violin Concerto in D Major, Op. 35. **Igor Stravinsky:** Violin Concerto in D. Jean-Jacques Kantorow, violin; London Philharmonic Orchestra, Bryden Thomson, conductor. 81757 3325 2 (1987).*

Another worthwhile release I neglected when it came out. Kantorow is a brilliant French violinist with a leaning toward fast tempi and sensitive, elegant playing rather than the big line. The neoclassical, superbly witty Stravinsky concerto from the composer's middle period suits that approach perfectly; the Tchaikovsky could be more heart-on-sleeve, but then it's schmaltzy enough when played with restraint. (And my taste is vulgar enough so that I love it, no matter what!) The orchestral framework by Thomson and the LPO is highly satisfactory, and the sound is vintage Kawaguchi.

***Peter Ilyich Tchaikovsky:** Symphony No. 5 in E Minor, Op. 64. **Boris Blacher:** Orchestervariationen, Op. 26. Frankfurt Radio Symphony Orchestra, Eliahu Inbal, conductor. 81757 6364 2(1989).*
***Peter Ilyich Tchaikovsky:** Symphony No. 6 in B Minor, Op. 74 ("Pathétique"). **Richard Wagner:** Vorspiel und Liebestod, "Tristan und Isolde." Franfurt Radio Symphony Orchestra, Eliahu Inbal, conductor. 817579715 2(1991).*

The two Tchaikovsky performances, recorded two years apart, show a slight shift by Inbal over that period toward a livelier, more dramatic approach. Both interpretations are in his deliberate, carefully detailed, often illuminative, and in the final analysis un-Russian style, but the *Pathétique* has greater impact. It also happens to be a much better recording, one of Denon's best, considerably smoother in the climaxes than the older CD. The umpteenth set of variations on the famous Paganini Caprice by Boris Blacher (1903-75) is not the best (nor the second best) set; as for the Wagner, it's lovingly shaped and tonally beautiful, but the seething intensity is missing. (Listen to Toscanini's 1952 recording and you'll know what I mean.)

dmp

Tom Jung is the undisputed master of New York studio-style recording; his CDs of small and medium-sized groups on his dmp (Digital Music Products) label fairly leap out of the speakers, with immense realism, very high definition, tremendous dynamics, yet no obvious gimmickry. You couldn't ask for better audiophile demo material. Not surprisingly, Tom has been using 19-bit and 20-bit A/D conversion for years now to make dmp's master tapes.

"Different Strokes." The Robert Hohner Percussion Ensemble. CD-485 (1991).

I think the idea here was to make the "ultimate" percussion recording, and in my opinion the effort was successful. If you have big speakers and lots of watts, you must get this CD—it will clean out your ears better than Q-tips. Not that it's a lowbrow audiogoon special, far from it. How could it be with the music of Darius Milhaud (on five marimbas, yet!), John Cage, Christopher Rouse, and other good 20th-century composers? Some of the sonorities are delicate, some are overpowering, and the playing by the 17 percussionists is as musical as you could ask for. The fidelity on a scale of 1 to 10 is an 11. Recommended to music lovers, not just drum freaks.

Dorian

Some would argue that this is *the* label for audio quality; my perception is that their best work is absolutely unbeatable but that their sound is just a little idiosyncratic in a few releases. Craig Dory, one of the founders and chief recording engineer of Dorian, is a relentless perfectionist with a very keen and critical ear; he is also a curious combination of highly credentialed technologist and tweaky audio fantast. He obviously doesn't like a close-up, dry sound (I do for certain kinds of music), preferring a considerable amount of ambience in his recordings. In the Troy Savings Bank Music Hall he sometimes goes overboard in his celebration of the great hall sound, but in the marvelous Eugene McDermott Hall of the Morton H. Meyerson Symphony Center in Dallas he achieves the current pinnacle of digitally recorded orchestral sound. The producers are not always the same in Dorian recordings, and lately Craig Dory has been delegating the engineering of lighter sessions to his associates. Coming Dorian releases will originate from 20-bit A/D coded masters.

***J. S. Bach:** "The Organ Works of Johann Sebastian Bach," Volumes 2 through 5. Jean Guillou on the Great Kleuker-Steinmeyer organ of the Tonhalle, Zürich. DOR-90149/50/51/52 (1990).*

After the initial Volume 1 recorded in 1987 at the Église Notre-Dame des Neiges in Alpe d'Huez, France (see my rave review in Issue No. 13), this monumental project moved to Switzerland, where the nearly five hours of music on these four CDs were all recorded in a marathon session in August 1990, then released piecemeal. The producer, Randall Fostvedt, still speaks in awed tones about Jean Guillou's nonstop feat of musical endurance. The Zürich Tonhalle organ is a much larger and more imposing instrument than the one at the French ski resort, but both were designed by Guillou and share some of the same irresistible tonal qualities. The big toccatas, preludes, fugues, and other major pieces in the Bach organ canon fare particularly well on the larger instrument, but the smaller-scale works also sound wonderful because the polyphony remains crystal-clear regardless of the nature of the music. As for the performances, all organists I am familiar with sound boring next to Guillou, who may not be most literally accurate and respectful Bach interpreter, just the most musical and captivating. He proves that these icons of polyphonic composition can be fresh and exciting. The recorded sound is simply the best there is in organ music; the same crew in the same place was responsible two years earlier for the Mussorgsky-Guillou "Pictures" that the hippest audiophiles still use as a test record.

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*"An American Panorama." **Leonard Bernstein:** On the Waterfront. **Roy Harris:** Symphony No. 3. **Aaron Copland:** Billy the Kid. Dallas Symphony Orchestra, Eduardo Mata, conductor. DOR-90170(1992).*

I compared this version of *Billy the Kid* with the Schwarz/Seattle recording on Delos, and it's a tough choice. Mata has the better hall, by far, and the better orchestra, by a narrower margin. Schwarz has a better understanding of the Copland idiom—a much snappier, jazzier performance. Mata is too old-world and careful (as he is in the other two American works). Both recordings are superb, the Dorian a little smoother overall and cleaner in the climaxes, the Delos more close-up and explosively dynamic. It's a nice dilemma. As for Bernstein and Harris—superior movie music and somewhat arid "serious" music, respectively—Mata doesn't quite have his heart in the performances.

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***Johannes Brahms:** Piano Trio No. 1 in B Major, Op. 8. **Antonín Dvořák:** Piano Trio in E Minor, Op. 90 ("Dumky"). The Rembrandt Trio: Valerie Tryon, piano; Gerard Kantarjian, violin; Coenraad Bloemendal, cello. DOR-90160 (1991).*

My personal prejudice: most of Brahms's chamber music is boring; most of Dvorák's chamber music is delightful. (So sue me.) I find this pairing no exception, but the playing by the Canadian trio is first-rate throughout. The recording is quite lovely but would be even better without the far too reverberant Troy hall signature (and Craig Dory wasn't even in on this one).

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***Antonín Dvorák:** "Complete Music for Violin and Piano." Ivan Zenaty, violin; Antonin Kubalek, piano. DOR-90171 (1992).*

This is one of the most natural-sounding recordings of the violin I know of, again a Craig-Dory-less recent production in the Troy hall. The piano is a little remote but the liveness is just right. The program is all of the 75 minutes of music Dvorák composed specifically for violin and piano; none of it is great but all of it is lovely. The two Czechs play it *con amore;* it's pure joy from beginning to end.

***Joseph Haydn:** Symphony No. 82 in C Major ("The Bear"); Symphony No. 38 in C Major; Symphony No. 104 in D Major ("London"). Los Angeles Chamber Orchestra, Christof Perick, conductor. DOR-90168 (1992).*

The immortal "London" receives a somewhat less than immortal performance here; it's merely well played—very carefully and lucidly—as are the earlier symphonies. The recording, however, proves that Craig Dory tends to hit a home run when he comes to a new venue (in this case Royce Hall on the UCLA Campus, Los Angeles) and only later starts to tweak around with the sound. Here the sound is all creamy goodness, transparency, and just-right ambience. Please do it exactly the same way next time, Craig!

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***Joseph Haydn:** Piano Trios. Trio in E Minor, Hob. XV: 12; Trio in C Major, Hob. XV:27; Trio in E Major, Hob. XV:28; Trio in G Major, Hob. XV:25. Kalichstein-Laredo-Robinson Trio: Joseph Kalichstein, piano; Jaime Laredo, violin; Sharon, Robinson, cello. DOR-90164 (1991).*

Wonderful music, wonderful playing by a world-class trio, wonderful recording by a Dorian team minus Dory. I could listen to this all day long; each trio, each movement is better than the last, and much of it seldom performed. These three superb musicians play as one, and the Troy hall is being used correctly here, as a framework, not as a feature. You have to be tone-deaf not to love this CD.

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***Franz Schubert:** Winterreise, D. 911. Victor Braun, baritone; Antonin Kubalek, piano. DOR-90145 (1990).*

If Schubert was the

greatest composer of lieder, as he almost surely was, and if *Winterreise* is his masterpiece, as it is widely considered to be, then if you have only one lieder cycle in your collection, this should be it. The question is, should it be Victor Braun's performance? Well, he has a beautiful voice, very strong and free on top, and he sings most expressively, with excellent musicianship and without any mannerisms. Fischer-Dieskau, for one, has more "personality," but that isn't necessarily the definitive way of singing this. Kubalek's sensitive accompaniments are another point in favor of this recording, which has my highest recommendation. It would have made me even happier if Craig Dory had moved in a little closer on this intimate music and kept the swimmy Troy hall ambience out of it, but *c 'est la vie.*

***Dmitri Shostakovich:** Symphony No. 7 in C Major, Op. 60 ("Leningrad"). Dallas Symphony Orchestra, Eduardo Mata, conductor. DOR-90161 (1991).*

I think this music is of quite limited artistic value, possibly the composer's weakest major work, although it served its purpose as patriotic propaganda in 1942. Half a century later, with even the name Leningrad gone, all that is left is that insufferably long and banal first movement and then more in the same vein. This performance is as good as I have ever heard, vigorous, precise, with great playing by the brasses, and the recording is stunning, confirming the status of the Craig Dory sound in the Dallas hall as one of today's major audiophile experiences.

Erato

This distinguished and at one time independent European label originated in France but is now, to the best of my knowledge, part of the Time Warner empire, tied in with Teldec. The producers varied in the Chicago recordings discussed below, but the recording engineer in each case was Larry Rock (great name for a classical recordist). It should be mentioned that Daniel Barenboim took command of the great Chicago orchestra only a few months before these recordings were made.

***Gustav Mahler:** Das Lied von der Erde. Waltraud Meier, mezzo-soprano; Siegfried Jerusalem, tenor; Chicago Symphony Orchestra, Daniel Barenboim, conductor. 2292-45624-2 (1991).*

With better singing this might have been a very satisfying performance of Mahler's crowning masterwork because Barenboim understands the idiom and the orchestra plays magnificently. Jerusalem, however, screams instead of singing when the going gets tough, and Meier is just mediocre. The recording is very good though a bit harsh in the climaxes.

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***Maurice Ravel:** Daphnis et Chloé (Suite No. 2); Rapsodie espagnole; Pavane pour une infante défunte; Alborada del gracioso; Boléro. Chicago Symphony Orchestra, Daniel Barenboim, conductor. 2292-45766-2 (1991).*

With an orchestrator like Ravel and an orchestra like the Chicago, you can't miss if you just play all his notes, and Barenboim certainly does that, with a great deal of control and beauty of sound. That extra magic, however (the Pierre Monteux kind), just isn't there. The recording is great, the cleanest and most spacious of the Chicago releases on this label so far, with a particularly nice bass drum.

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***Richard Wagner:** Der Ring des Nibelungen (excerpts). Deborah Polaski, soprano; Chicago Symphony Orchestra, Daniel Barenboim, conductor. 2292-45786-2 (1991).*

I grew up on Toscanini's and Szell's Wagner, and currently I favor James Levine. This just isn't as exciting. The five excerpts *(Walkürenritt, Waldweben, Rheinfahrt, Siegfrieds Tod,* and the final immolation scene) are played with great clarity and gorgeous orchestral sound, but the grandeur is missing and Polaski is a highly forgettable Brünnhilde. The recording is excellent overall but again a little strained on brassy climaxes.

Harmonia Mundi

This is a French label, but their scope is international and their USA affiliate distributes many more small international labels. The audio quality of their recordings has been variable; those made by Peter McGrath are generally superb.

***George Frideric Handel:** Messiah. Six soloists; U.C. Berkeley Chamber Chorus; Philharmonia Baroque Orchestra, Nicholas McGegan, conductor. HMU 907050-52 (1991).*

Handel changed the score of *Messiah* many times to suit available singers, and this is the first and only recording that offers every note of all the different versions, with programmable indexing. The performance is of course superauthentic à la McGegan; the orchestral playing is wonderful; the chorus is weak; the soloists are acceptable to pretty good; the Peter McGrath recording is marvelously lucid.

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***George Frideric Handel:** Giulio Cesare. Jennifer Larmore, Barbara Schlick, et al., Concerto Köln, René Jacobs, conductor. HMC 901385-87 (1991).*

If you get to know only one Handel opera, this should be the one, his best, with many truly brilliant moments. This is a genuinely enjoyable but very far from great period-practice performance; the same can be said of the recording, which is clean and transparent but often thin and too bright. One big point for Handelians: no cuts.

London

Of all the big international labels using more or less standard multimiking techniques, London (i.e., English Decca) comes closest to what could be called an audiophile sound. The recording engineer responsible for that is usually John Dunkerley, Colin Moorfoot, or John Pellowe.

***Hector Berlioz:** Symphonie fantastique; L'Invitation à la valse (Weber, arr. Berlioz). The Cleveland Orchestra, Christoph von Dohnányi, conductor. 430 201-2 (1989).*

Flabbergastingly brilliant playing by the Clevelanders that just bowls me over—what a band!—but Dohnányi is a bit too stiff and controlled in this music, perhaps because he wants to set some kind of new record in precision. The sound, recorded by John Pellowe in the Masonic Auditorium, is wonderfully detailed down to the lowest notes, as good as a Decca can get.

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***Robert Schumann:** Symphony No. 1 in B-flat Major, Op. 38 ("Spring"); Symphony No. 4 in D Minor, Op. 120. Royal Concertgebouw Orchestra, Riccardo Chailly, conductor. 425 608-2 (1988).*

No. 4 is the better work in my opinion, with some beautiful passages, but neither one is a truly great symphony. I prefer Schumann as a composer for the piano. Chailly has no special feeling for these works, either; he conducts them with his usual vigor and that's that. The Amsterdammers play with the utmost virtuosity and beauty of tone, as always; the recorded sound is round and clean but could be more transparent (probably Schumann's fault, not the Decca engineers').

Marco Polo

This European "label of discovery" (as they call themselves) is distributed in this country by Harmonia Mundi USA; the CDs are manufactured in Germany.

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***Havergal Brian:** Symphony No. 1 ("The Gothic"). Four soloists; seven choruses; CSR Symphony (Bratislava) and Slovak Philharmonic, Ondrej Lenard, conductor. 8.223280-281 (1989).*

Havergal Brian, an Englishman, died in 1972 at the age of 96 and left a legacy of 32 symphonies, hardly ever performed. This, his first, is something of a cult item because of the gigantic forces needed to perform it; the orchestra alone requires almost 200 players, and then there are the huge double choruses, soloists, etc. (You can look it up in *The Guinness Book of Records.* No kidding.) This recording evokes Samuel Johnson's "like a dog's walking on his hinder legs—it is not done well, but you are surprised to find it done at all." Not that it's done badly, far from it, but I think only a Solti/Chicago type of performance of a bloated and somewhat incoherent, episodic work like this could give it genuine shape and thrust, and this is not quite on that level, although the playing is highly professional. As an audiophilic curiosity, however (the Strauss/Mahler/Schönberg sound cubed), I can recommend this recording because the sound is amazingly good—transparent and panoramic, with a wide dynamic range, clean highs, strong bass, and only a few moments of strain.

MusicMasters Classics

An affiliate of the BMG group (which now also includes RCA Victor), this label has an international stable of top artists and works with world-class producers and recordists like Max Wilcox.

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***J. S. Bach:** Goldberg Variations. Vladimir Feltsman, piano. 01612 - 67093 - 2 (1991).*

Recorded live in Moscow, this performance is just a few seconds short of 80 minutes because every repeat is observed. To avoid tediousness, Feltsman jazzes up the repeats by playing them in a different register, switching the voices by crossing hands, etc. Some will be offended by this; I am not, but I still like Glenn Gould's Bach a lot better. Feltsman has the chops but doesn't have major insights to offer. The sound is basically good but a little dry and thin, with some audible coughs in the hall.

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***Igor Stravinsky:** Pulcinella Suite; Symphony in C; Russian Peasant Choruses; Russian Sacred Choruses; Les Noces. The Orchestra of St. Luke's & The Gregg Smith Singers, Robert Craft, conductor. 01612-67086-2 (1992).*

All of this is first-chop middle-period Stravinsky, pungent and wonderfully listenable. It is Volume II of the complete Stravinsky canon being recorded by Robert Craft, who as the composer's closest associate is entitled to whatever performance style he sees fit. For my musical palate he is a little *sec*—in the *Pulcinella* he sounds like a Nicholas McGegan on steroids demonstrating period-practice Frescobaldi—but the orchestra is very good, the singers are very good, Craft's grasp of the music is unquestionably of a high order, and the whole thing works as a musical experience. The recording is extremely lucid and clean, just like the playing.

RCA Victor Red Seal

The granddaddy of all labels is now part of the BMG Music empire—nothing is sacred anymore. Their sound quality has been good lately but basically middle-of-the-road.

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***Antonín Dvorák:** Piano Quintet in A Major, Op. 81; Piano Quintet in A Major, Op. 5. Rudolf Firkusny, piano; Ridge String Quartet. 09026-60436-2 (1990).*

***Antonín Dvorák:** Piano*

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Concerto in G Minor, Op. 33. ***Leos Janácek:*** *Concertino; Capriccio. Rudolf Firkusny, piano; Czech Philharmonic Orchestra, Václav Neumann, conductor. 09026-60781-2 (1990-91).*

Dvorák and Firkusny go together like ham and eggs, like Shakespeare and Olivier. He was 78 and 79 when he recorded these works, but the lovely Op. 81 quintet and the almost as lovely piano concerto still sing with incomparable lyricism under those magic fingers. As for the quirky but fascinating Janácek pieces, he is *the* authority, and of course the Czech Philharmonic can't hurt in Czech music. The recorded sound is unspectacularly excellent in each case. Recommended!

Reference Recordings

This is the small independent label founded and run by Tarn Henderson; its audio engineering wizard is the legendary Keith ("Prof.") Johnson, whose best recordings are as good sonically as anything in the world and whose worst are still very respectable.

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Malcolm Arnold: *Overtures. The London Philharmonic Orchestra, Malcolm Arnold, conductor (David Nolan, leader). RR-48CD (1991).*

This is rather lightweight but colorful orchestral fare by England's most famous trumpeter, almost as famous film-music composer, and less famous symphonist. The recording, however, is simply staggering, possibly Keith Johnson's masterpiece and a contender for the title of Best Demo CD. Both the hall acoustics and the brilliant instrumental textures are sensational—and this is from before RR's switch to the 20-bit HDCD system. I won't dwell on it further; just go out and get it!

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"Fennell Favorites!" Frederick Fennell live *with the Dallas Wind Symphony. RR-43CD (1991).*
"Fiesta!" Dallas Wind Symphony, Howard Dunn, conductor. RR-38CD (1990).

The Meyerson/Dallas acoustic raises these terrific, in-your-face Keith Johnson band recordings to demo level. Excellent band; splendid Hispanic sonorities in "Fiesta!" (the bass drum is awesome); solid classics (Bach, Brahms, Prokofiev, etc.) transcribed for band on the Fennell CD.

Telarc

Michael Bishop has taken over some of Jack Renner's responsibilities as Telarc's recording engineer; the Levi/Atlanta recordings, for example, are entirely his currently. The Telarc sound I have praised is a house *cuvée*, however, and remains constant.

J. S. Bach: *"Switched-On Bach 2000." Wendy Carlos. CD-80323 (1992).*

As a 25th-anniversary sequel to the original Carlos electronic Bach album, this sounds perhaps a little less fresh and startling musically but a lot more sophisticated electronically. Check out the concluding Toccata and Fugue in D Minor for the ultimate in synthesized timbres for organ music. Soundwise, your system is the only limit here.

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Sir Edward Elgar: *Symphony No. 1 in A-flat Major, Op. 55; Pomp and Circumstance Marches, Op. 39. Baltimore Symphony Orchestra, David Tinman, conductor. CD-80310 (1991).*

Every audiophile needs a blockbuster version of Pomp and Circumstance March No. 1, and this does the job in spades. The aesthetic fallout of such a crass pursuit is a beautifully shaped, fine-sounding performance of the much more important Symphony No. 1. This is a good orchestra, no doubt about it.

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George Frideric Handel: *Messiah. Four soloists; Boston Baroque Orchestra & Chorus, Martin Pearlman, music dir. & harpsichordist. 2CD-80322 (1992).*

When it rains it pours; here's another fine period-instrument, period-practice performance of *Messiah*, perhaps even better than the one on Harmonia Mundi but offering only the standard version without the extra goodies. The chorus is clearly better; the singers are somewhat better; the conducting isn't better; and the recording is more conventionally creamy-rich in texture.

Bedrich Smetana: *Má Vlast (My Country). Milwaukee Symphony Orchestra, Zdenek Macal, conductor. CD-80265 (1991).*

That this is the sonically best-ever recording of this beautiful and colorful cycle of tone poems is almost indisputable. Jack Renner outdid himself; the sound is rich, suave, and powerful beyond belief. The performance is arguable. Some would surely prefer a more lyrical, more heart-on-sleeve, more typically Czechoschmaltzian approach. Macal's interpretation is warm yet strong and firmly structured, not at at all sentimental. I buy it; you may not. But his name is Zdenek and yours is probably Bob.

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Igor Stravinsky: *Le Sacre du printemps (The Rite of Spring); Pulcinella Suite. Atlanta Symphony Orchestra, Yoel Levi, conductor. CD-80266(1991).*

Another title contender for Best Demo CD. The dynamic range and the accuracy of instrumental delineation are just amazing. The various drum explosions in the *Sacre* are what audiophiles are willing to die for. The Atlanta orchestra's playing under Levi is extremely disciplined, controlled, precise; the pacing is good in the Sacre; and the *Pulcinella* is equally well played.

Teldec

The star label of the Time Warner empire.

Ralph Vaughan Williams: *Symphony No. 6 in E Minor; Fantasia on a Theme by Thomas Tallis; The Lark Ascending. BBC Symphony Orchestra, Andrew Davis, conductor. 9031-73127-2 (1990).*

Truly great, profound music, very knowledgeably performed and well recorded (for a big company).

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