Issue No. 17 Winter 1991-92

The Audio Critic®

Retail price: $7.50

In this issue:

More loudspeaker reviews, including a first look at the remarkable new Win SM-10 Broadcast Monitor.

The promised survey article on the various different approaches to bass optimization and compact subwoofer design makes its delayed appearance.

Our highly popular expose of the wire/cable scene advances to the brutally simple subject of interconnects.

Reviews of highly advanced multibit D/A processors, Dolby S cassette decks, and a high-end TV monitor.

Plus all our regular columns and features, an expanded CD review section, and two special reports on the clash of science and voodoo at an unusual AES convention.

Issue No. 17 **Winter 1991-92**

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For subscription information and rates, see inside back cover.

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Editor/Publisher's Note

Well, what did I tell you? This is the Winter 1991-92 issue and it's still winter as you're reading this, right? Not exactly early winter, as planned, but winter nonetheless (the last day of winter being March 19 this year). The previous issue came out in the fall, and the Spring 1992 issue will likewise be published in the spring, so it appears that this so-called quarterly is now actually on a quarterly schedule—sort of. My plan is to make it a bimonthly (six times a year) as soon as possible, but first we must thoroughly solidify the quarterly timetable.

Regularity is part of professionalism in magazine publishing; unfortunately it doesn't assure professionalism in equipment testing and editorial practice, as exemplified by certain regularly published audio periodicals. My aim is regularity without any compromise in quality and, especially, without shooting from the hip under deadline pressures. It's hard but not impossible.

* * *

You will notice an emphasis on loudspeakers in this issue, reflecting the priorities of my audio philosophy. I believe that the audiophile who is well-informed about loudspeakers will end up with a much better-sounding system than the hairsplitting amplifier tweak (not to mention the cable cultist and related mystics). Realistic assessments of audio electronics will continue to appear in our pages, but you can expect loudspeakers to receive top billing.

Box 978

Letters to the Editor

*We have many thousands of new subscribers as a result of our recent promotional efforts, and this box is overflowing, mostly with letters of praise from audiophiles to whom **The** Audio Critic is a new experience. They all seem to have gotten together and agreed on the phrase "a breath of fresh air." Like most love letters, these are heartwarming—thanks!— but editorially uninteresting. 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:

I read your recent article "The Wire and Cable Scene: Facts, Fictions, and Frauds, Part II" with interest. While I applaud your attempt to strip away the "snake oil" and pseudotechnical "BS" in an area fraught with it, I would like to offer the following items for your consideration. First, before I start, I would like to say that I do have a very slight commercial interest in well-designed audio/video cables for the studio/home for whatever bias this may bring to my arguments. In addition, I am responsible in my main employment for the design of VLSI devices and leading-edge digital/analog systems for telephony. Second, I would like to say that my points apply only to the measurable issues associated with speaker cable; I will let someone else draw the line between where audiophile perception ends and bats' & dogs' perceptions continue.

Frequency response variations, due to the effect of series resistance and inductance on an amplifier's ability to control its load under small-signal conditions with increasing (audio) frequencies, is just one area of legitimate speaker cable design. Fortunately, lowering the series resistance and inductance is simple, relatively inexpensive, and has no significant drawbacks, provided that the amplifier in question has adequate phase margin.

Control of a load by the amplifier under large-signal conditions is another important design aspect [1]. Data shows that real-world current requirements for amplifiers and the associated speaker cable can run 3-6 times that required for an equivalent resistive load for multiple-driver loudspeaker systems [2]. The solution again is to lower the series inductance and resistance.

As you point out, ideally an amplifier should appear like an "almost perfect voltage source" to the load. Not only do real-world amplifiers have some finite output impedance, but the speaker cable adds substantially to this impedance. Once again, reducing inductance and resistance are the answer to lowering this impedance. At the same time, reducing this impedance also decreases interface intermodulation distortion in the amplifier [3], [4], [5].

Another widely ignored, measurable problem is the VLF transmitter and antenna loop formed by the amplifier, cable, and load. In a system with the voltage gain involved for phono reproduction, minimizing the antenna loop area formed by the speaker cable by using short, closely-spaced (low-inductance) cable will definitely produce a measurable improvement [6]. With some commercially available cables, noise induced into a phono "front end" is only -45 dBV at the output of the line-level preamplifier!

Maintenance of an amplifier's output common-mode rejection via the use of a cable employing "identical" conductors to connect to each side of the load is also an important issue [7]. This criterion is slightly more difficult to achieve simultaneously with low inductance. Coaxial cables do not have adequately matched conductors unless two runs, physically tied together, are used with the second run having the conductors reversed. In effect, the center conductor of one cable and the shield from the other are wired in parallel to connect to each side of the load; this is sometimes referred to as "reverse biwiring."

Finally, the dielectric and physical characteristics of cable construction are important, as they affect capacitance changes over frequency, level, and vibration. Dielectric issues can be minimized by using a low-dielectric-constant material like DuPont Kapton, polypropylene, or DuPont Teflon. The use of a tight-fitting, fairly rigid cable jacket will further minimize any physical problems like microphonics.

Length, as you stated, is important as it affects each of the areas described above.

While I agree with the intent and some of the substance of both your article and

Dr. R. A. Greiner's articles, I find them both limited in scope. I also find, given the probable audience and the equipment that they are likely using, that some of your conclusions are oversimplifications, for four reasons. First, commercial cables that perform well (significantly better than zip cord) in most of the respects described above are available, and are inexpensive relative to equipment costs, e.g., Hitachi's coaxial speaker cable, Mogami (Boulder) 2477, and the lower-cost Straight Wire coaxial cables like Teflon 12 (although some of these may not be current model designations). Second, some users have electronic biamplified systems from which they have endeavored to remove fuses, passive crossover components, and other links that tend to minimize the measurable benefits of better speaker cable, and they may want to know what cables are available that maximize the measured performance of their systems irrespective of price. Third, many readers will want to know what some of the valid criteria are, especially in the presence of so much "hype" in this area. Last, via hearsay, I understand that you do not use zip cord, but rather Mogami 2477 (Boulder), so why recommend zip cord to the customers of your periodical?

In closing, I want to wish you continued success and add another accolade for Dr. David Rich's article on CD player technology.

Sincerely yours,
David S. Mohler
Westminster, CO

References

[1] R. R. Cordell, "A MOSFET Power Amplifier with Error Correction," *Journal of the Audio Engineering Society* 32 (January/February 1984): 2-17.

[2] I. Martikainen, A. Varla, and M. Otala, "Input Current Requirements of High-Quality Loudspeaker Systems," presented at the 73rd Convention of the AES, *Journal of the Audio Engineering Society (Abstracts)* 31 (May 1983): 364 (Preprint 1987).

[3] R. R. Cordell, "Open-Loop Output Impedance and Interface Intermodulation Distortion in Audio Power Amplifiers," presented at the 64th Convention of the AES, *Journal of the Audio Engineering Society (Abstracts)* 27 (December 1979): 1022 (Preprint 1537).

[4] E. M. Cherry and G. K. Cambrell, "Output Resistance and Intermodulation Distortion of Feedback Amplifiers," *Journal of the Audio Engineering Society* 30 (April 1982): 178-91.

[5] M. Otala and J. Lammasniemi, "Intermodulation Distortion in the Amplifier-Loudspeaker Interface," presented at the 59th Convention of the AES, *Journal of the Audio Engineering Society (Abstracts)* 26 (May 1978): 382 (Preprint 1336).

[6] D. S. Mohler, "Improving the Amplifier-Loudspeaker Interface," AT&T Bell Labs, Denver Audio Club, September 1989.

[7] S. Takahashi and S. Tanaka, "A Measurement Method of Hum Modulation Caused by a Loudspeaker's Electromotive Force," presented at the 70th Convention of the AES, *Journal of the Audio Engineering Society (Abstracts)* 29 (December 1981): 939-40 (Preprint 1823).

I'm somewhat bewildered by this letter, which is 65% solid science, 20% audiophile angst, and 15% cloud-cuckoo-land.

I totally agree with the points referenced with [1] through [5] and [7]. My speaker cable article also shows that minimizing R and L yields the best response, and I favor the shortest possible cable for that reason. I have nothing against your double run of coaxial cable, either, although it's probably overkill. That the speaker cable can be part of an "antenna" that transmits noise into a high-gain low-level stage is an interesting idea that I find very plausible even though I never had occasion to deal with it. I note that you reference your own work [6] on this subject.

But "dielectric issues"? Vibration? Microphonics? In speaker cables? At audio frequencies? Here you cross over into Enid Lumley country—and of course run out of references. Where are the AES papers on tight-fitting, rigid cable jackets, etc., etc.?

As for my recommendations and what I personally use, I think you're just quibbling. I mentioned zip cord to drive home the point that for a connection of, say, four feet or so the kind of wire doesn't matter; you could use a wire coat hanger with the contact points scraped For long runs I did recommend coaxial cable of sufficient gauge, just as you do. Yes, I own two long runs of Mogami Neglex 2477, which I originally obtained from Boulder (see Issue No. 10, page 22—no hearsay!), but currently I'm using very short lengths of nameless 14-gauge two-conductor cable. None of the above is politically correct to uptight high-enders, of course, but that shouldn't bother a technologist like you. Lastly, I don't quite understand your points about maximizing "the measured performance" and about "the valid criteria"—didn't my article address exactly those issues?

Maybe a more detailed explanation of your "very slight commercial interest" would shed some light on the strange incongruities of your letter. In any event, thank you for the positive comments.

—Ed.

The Audio Critic:

I got a real kick out of your article on p. 51 [of Issue No. 16], "The Wire and Cable Scene: Facts, Fictions, and Frauds, Part II."...It took some guts to write that. It won't make you many friends; neither the highfalutin cable manufacturers nor the customers whose illusions you smashed will like you for it.

Oh yes, in thumbing through, I just came across "Hip Boots," where you dumped on George Tice. Great job! I agree with you totally on both speaker cables and injecting clock pulses into the line. We use 18-gauge zip cord (lamp cord) between our amps and speakers. By the way, I have a graduate degree in electronics, an MS in EE.

One point I would like to emphasize, though—*there is no $1200 amp that can drive some of the more difficult speakers, like the big Apogees, Infinities, Duntechs, etc. For this you need a $6000 amp.* The $1200 amp would go up in smoke.

Now you may say, "Who needs these monster speakers? A smaller, dynamic, high-efficiency speaker is good enough." Maybe—that is a subjective judgment. But the big speakers exist; a lot of people like them; and it takes an amp with a lot of balls to drive them.

Sincerely,
Jack Jones
President
NRG Control, Inc.
Walled Lake, MI

P.S. In our power amps we developed our own low-inductance cable to prevent ringing and instability. Low inductance is the key. We could not find anything satisfactory on the market.

I didn't "dump" on George Tice; I criticized, and protested against, the way he dumps his stuff on gullible audiophiles.

As for amplifiers, do you think the Adcom GFA-585 ($1200), the Carver TFM-45 ($949), or the Hafler XL-600 ($1299) "would go up in smoke" driving any speaker that doesn't drop below 2 ohms impedance at any frequency? I don't think so. And that covers the great majority of the monster speakers.

Your need for exceptionally low-inductance cable in your power amplifiers is due to their 1 MHz bandwidth. I'd like to

be shown that, say, 100 kHz is insufficient. I suspect that in this case one excess begets another.

Thanks for the compliment.

—Ed

The Audio Critic:

...The overall approach of your reviews appeals to me. I just can't be convinced that interconnects costing hundreds of dollars can make huge improvements in the sound. Otherwise, simple receivers should have a significant advantage over costly separates.

Thanks,
Richard L. Brown
West Hartford, CT

Very good point. Nothing can possibly be lower in distortion/coloration than hardwiring over a couple of centimeters without any transmission cable. In a receiver, the tuner section is directly wired in this manner into the preamplifier/control section, which in turn is similarly wired into the power-amplifier section. Therefore, as far as line-level connections are concerned—and nothing else—a receiver beats high-end separates hooked up with high-end interconnects because no cable can be quite as good as no cable at all.

—Ed

The Audio Critic:

Thank you for the issue [No. 16] of *The Audio Critic.* I couldn't put it down on my train ride...to New Haven! The quality and depth of the reviews were superb, the humor poignant. If George Tice meets your challenge I would check the credentials of his supporters very closely! I've read his "white paper" with some astonishment that he has the *chutzpah* to put so much garbage into print.

I especially liked your speaker cable simulations, for obvious reasons. The speaker model I used in my simulations was not a broadband model as yours, but rather a very simple single-frequency model. Would it be possible to obtain a copy of the "Amazing Loudspeaker" model you used?

Like many others before me, [I find that] letters to *Stereophile* have disappeared into a black hole. In my latest effort, I pointed out to them that the phase angle in their speaker impedance/phase plots has the polarity reversed! Readers should be aware **that** ***all of the phase plots to date are incorrect.*** A simple setup parameter on the Audio Precision would have corrected this.

Thank you again for your interest and kind words.

[Four weeks later:]

In my last letter, I mentioned the gross error in the speaker phase plots published **in *Stereophile.* Well, *Stereophile* has finally** acknowledged the error in their speaker phase plots, but just barely. It appeared in a footnote in the *third* speaker review in the November 1991 issue (p. 167). The first two reviews made no mention of any previous errors, even though the current plots were corrected. In the footnote, Mr. Atkinson says that only *some* of the plots are wrong. I can say without hesitation that *all* of the plots performed on the Audio Precision between May 1990 and October 1991 are incorrect. I can't believe they got it right once but not now. Only one plot that didn't use the Audio Precision seemed correct (B&W Matrix 800, June 1991), but it is hard to tell since the scale factor on the graph spanned 180° in inch. Mr. Atkinson also says how he's puzzled at how this could happen. I suggest he read the Audio Precision owner's manual. Perhaps it's never too late.

Sincerely,
Fred E. Davis
Hamden, CT

Coming from an electronics engineer and author of the excellent paper on loudspeaker cables in the June 1991 issue of the AES **Journal,** ***your favorable comments are especially welcome. I've received similar encouragement from a good many other professionals; indeed, it appears to be a nonnegotiable requirement of*** **hostility** ***to this publication*** **not** ***to have an education in physics, mathematics, or engineering.***

I'm reproducing the circuit of the Carver "Amazing Loudspeaker" Platinum Mark TV here in response to your request. I just didn't want readers to get involved in it and miss the main point of my article.

As for **Stereophile's** ***problems with the Audio Precision gear, it has always been my impression that they used it for cosmetic purposes—the "scientific" look-and-feel of their pages—rather than as an investigative tool.***

—Ed.

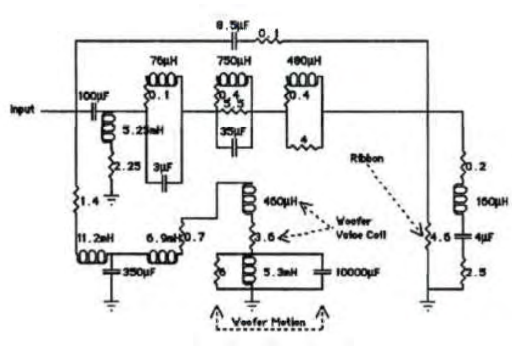
The Audio Critic:

I must admit that I really do enjoy your magazine. When I first got into stereo back in about 1979, I used to believe anything the stereo salesman told me. I was **reading a lot of *Stereo Review* and *High Fidelity*** magazines. The high-end stereo store that was in my town at that time would always tell me that I couldn't believe anything I read from these magazines (to include *Audio* magazine too!). They convinced me that magazines of this type accepted too much advertising to be impartial in their reviewing.

I would go to this high-end stereo store and wait with bated breath for the words of wisdom to spill forth from the stereo salesman's mouth. The main problem was that I could never really hear the great differences between amplifiers/preamplifiers/turntables (of course they only listened to "belt-driven" and never, never, never to "direct-drive") that I was supposed to hear. Of course, the stereo salesman could hear all of these supposed differences readily and was only too quick to point these differences out to me.

I don't know exactly when, but at one point I started asking myself why was I listening to this garbage that this supposed "expert" was putting forth. Did my ears not have a vote? Was my opinion not valid? Who made this "expert" the only vote in town?

After some real soul-searching I discovered that I could learn to trust my own judgment. I started reading about double-blind listening tests in *Stereo Review* and started to really wonder about the validity of some of the claims of my high-end stereo store. I also started to wonder about the validity of the high-end magazines. I could not hear the great, huge, unbelievable, etc., etc., differences that were purported to exist. At this point I decided that they probably did not exist (at least in the range of



Schematic of the loudspeaker system circuit used as the load in the computer simulations of speaker cable response in Issue No. 16. The various tuned circuits are for equalization (trimmers shown as single-value resistors).

human hearing capabilities). This is when I started to get smart.

I was extremely disappointed when *High Fidelity* magazine merged with *Stereo Review*, Another of the only rational stereo magazines gone and only the disappointing high-end stereo magazines to deal with.

When I first sent my money in to your company, I thought that you would be like so many other high-end stereo magazines that I've tried but just can't stomach (i.e., *TAS*, *Stereophile*, etc.). What I found instead (and much to my delight) was a rational, well thought-out, grounded-in-science magazine that relied on my favorite thing when in comes to evaluating stereo equipment: double-blind listening tests. This is where, in my opinion, the rubber meets the road. If these supposed differences are so great, then surely I or the high-end reviewers could hear them. Ha! Fat chance!

If a reviewer isn't willing to stand up to this kind of unbiased, scientifically set-up test, then in my opinion he's full of shit. (Sorry for the poor language but this is really how I feel.)

Again, thank you so much for being a good reviewer, as opposed to the English-major reviewer who doesn't really know what the hell he's doing but does know how to use adjectives. I'll take the double-blind reviewer every time! The other high-end magazines and stereo salesmen live off the insecurities of the nonsecure audiophile.

Your magazine is a refreshing breath of fresh air. Please [extend] my subscription.

Sincerely,
Robert L. Thompson
Fort Huachuca, AZ

P.S. Please write about yourself in your magazine in the future (i.e., how you got into audio, what schooling you have, etc.). I think people would be interested.

Your case history is instructive and heartening; unfortunately a lot of your fellow audiophiles haven't progressed beyond your initial phase and keep going back to that salesman for more of his voodoo.

As for my audio bio, let me just say that my hybrid schooling allows me to trade transfer functions with the engineers as well as adjectives with the English majors. Those who have read every issue of ***The Audio Critic*** *actually have a pretty complete idea of who I am and where I'm coming from.*

Your supportive comments are greatly appreciated, and your lapse in polite vocabulary is forgiven.

—Ed

The Audio Critic:

I've just finished digesting your introductory package of Issues No. 11 through No. 15. It would have been a bargain at twice the price.

I would like to express my interpretation of your view of the current amplifier/preamplifier scene to see if I understand you correctly.

What I think you're saying is this: If the given component is free of egregious engineering and manufacturing errors (given the current state of the art), then in all likelihood it will be audibly indistinguishable from a similar component, regardless of any price difference.

Or, to put it another way, if you were to drive, say, a pair of Vandersteen 2Ci speakers, and you were to use as amplifiers a Krell KSA-150 and a Harman/Kardon Citation 22, at normal listening levels a "typical" listener would be hard-pressed to find any major audible differences between the two. I understand that this would not necessarily apply with a much more difficult load, like an Apogee Scintilla, since the Krell has the ability to drive low-impedance loads without concern.

What I'm really trying to get at is this: The intelligent audio consumer will look closely at his main input sources, his listening room, his speakers, the types of music he usually listens to, and will be able to come up with a list of components that will satisfy his requirements. Then he would look at his monetary restraints and come up with a budget. After all this he will still be looking at a large group of manufacturers, from Adcom, Aragon, and Carver, to Rotel, Sumo, and Tandberg (to name just a few). But the point is that he will in all probability be just as pleased with the sonic performance of any of these brands, and would logically base his decision on price, availability, warranty, service considerations, etc. (given the caveat that you haven't actually tested any of these components, and that there are no compatibility problems, like an amplifier with a power supply/output capability insufficient for the load it has to drive).

On the lighter side, I've come up with a quick and easy way to differentiate your publication from the others. Here it is: Wide-eyed audio "enthusiast" writes to three audio magazines. He writes: "I've just put one of those magic bricks on top of my amplifier, and the sound has improved 100 percent!" The commercial hi-fi audio mag responds: "Wow! There must be something really wrong with your amp!" The "alternative" audio press would respond: "Wow! Another breakthrough. This one we've got to try!" Whereas the most logical (if not polite) response to such an affirmation would be: "Wow! There must be something wrong with this guy!" Can you guess who's going to pick number 3?

Please renew my subscription. Keep up the good work!

Sincerely,
Joel Ellingsworth
Austin, TX

Your exegesis of my electronic Weltanschauung *is essentially correct, but you leave out an important criterion on which the choice between soundalikes can be based: ergonomics. Some equipment is much easier and more pleasant to use than others; the human engineering is so much better. That would influence me more than, say, a 3-year versus a 1-year warranty.*

I realize that your carefully constructed little joke can't be rewritten without ruining your punch line, but your three responses are somewhat off the mark. The large-circulation hi-fi slicks would politely mutter something to the effect that they haven't had the same experience as the letter writer. The alternative audio press would probably try to one-up the writer by bringing up magnetic *magic bricks—or something. And I never say "Wow!" unless I'm wowed.*

—Ed.

The Audio Critic:

I [was] pleased to take you up on your subscription offer... I am a longtime subscriber to *Stereophile* and had already concluded that I could no longer stomach their antirational stance, although they have brought to my attention many fine products. I think that high-end audio is made to appear ridiculous by its cultists, and I am amazed by the self-delusion that prevails. Subjective reviewing is not in itself objectionable, but in the hands of the cultists it supports a rigid hierarchy of manufacturers who successfully cater to the obsessions of reviewers and who in turn are manipulated by the manufacturers. The exodus into the cloud-cuckoo-land of line conditioners, CD tweaks, and exotic cables was the last straw as far as I am concerned. I have been an audio hobbyist since the mid-1960s and, although not an engineer, I am at least scientifically literate. I also am a psychoanalyst and I think I know self-deception and obsessive rationalizing. I agree that this is done unintentionally and out of self-aggrandizement rather than simple greed, but it is a great disservice to those who expect some kind of substantive information from audio publications.

Manufacturers and retailers should be aware of the effect that this behavior has on consumers; in my own case I have delayed purchases because of the lack of reliable, credible information. I am reluctant to refer nonaudiophile friends who want something better than department-store brown goods to audio salons where they will get the party line about directional cables and CD stabilizers. The largest untapped market for home audio is among women, and they are having nothing to do with the bilge that comes from typical high-end dealers. They may or may not know physics, but they know when men are kidding themselves. Even the best dealers think they have to kowtow to the superstitions of the cultists; those who do not are referred to contemptuously as mid-fi appliance hawkers. Great way to promote a love of high-quality reproduced music: adopt a sneering attitude toward neophytes and nontweak products.

I wish you every success with the reincarnation of *The Audio Critic.* I am glad to find a publication that only embarrasses me occasionally (with a little bit of bombast to make a point) instead of constantly (with flaky, "politically correct" pronouncements about things that exist only in the reviewers' fantasies). I look forward to retrieving this most enjoyable hobby from the audio Moonies.

Sincerely,
Michael L. Pipkin, M.D.
Houston, TX

Your comments are totally on target; I'll take exception to two minor details only.

1. Women are probably not "the largest untapped market for home audio," although it would seem logical to think so. Only 3% of the subscribers to American Record Guide *are women, and the magazine's editorial content is all CD reviews and no audio. No untapped female CD-player buyers there. How that jibes with the frequent preponderance of women at concerts I don't quite understand myself, but I wouldn't put my life's savings into a ladies' audio department franchise.*

2. Verbum sat sapienti*—a word to the wise is sufficient, but bombast just barely gets the attention of the unwise.*

-Ed.

The Audio Critic:

Today I received a sample copy of *The Audio Critic.* Because I have a background in physics and biochemistry I lean towards an objective approach for the evaluation of high-fidelity equipment. However, I have been interested in the subject of good music reproduction since 1952 and I know that many of the qualities of sound reproduction equipment cannot be explained by currently used measurements. I also know that the only way to properly carry out an A to B comparison is to first educate the listeners on how to listen, something I gather you do not do. Nevertheless, I thought that your magazine would be useful and interesting. I was prepared to subscribe, but then I began reading your descriptions of your fellow writers in the field as "...self-indulgent, posturing little people... protecting the belief system of the cult...aren't big enough to admit they were wrong..." and similar scurrilous and derogatory statements scattered all through the journal. I realized that anyone who would write that way could not be objective or scientific in his evaluation of anyone or anything.

Unhappily yours,
Melvin L. Goldberg, M.D., Ph.D.
Altamonte Springs, FL

Since you haven't subscribed, you're unlikely to see my reply, but your letter typifies certain attitudes that I want other readers to recognize for what they are. You're wrong on three counts:

1. Anything that can be heard can also be measured, but the measurement protocol must suit the nature of the audible phenomenon. It's the routine measurements that sometimes leave us without an explanation.

2.1 only use educated ears, i.e., highly motivated and experienced audiophiles and music lovers—some of them professional musicians—in my listening tests. Whatever made you "gather" the contrary?

3. Strong opinions about other practitioners—including suspicions of small-mindednes and bad faith—are absolutely unrelated to objectivity or lack thereof in scientific inquiry. Are you aware of the opinions of Dr. Edward Teller (who comes from the same culture as I, a generation before me) about some of the other figures in the nuclear community? Is he incapable of correct scientific evaluations? If everyone shared your distaste for outspokenness and confrontation, all productive dialogue in our society would die of terminal blandness.

—Ed

The Audio Critic:

Weren't you stunned by John Atkinson's implication *(Stereophile,* August 1991, "Industry Update"/Australia) that the Garrott brothers and their wives committed suicide because of the commercial success of the Compact Disc? If only John Atkinson and Harry Pearson would take their analog love affair this seriously!!

The Garrott brothers and their wives had to be sickies, and the real tragedy is that these four lives couldn't be given to four terminally ill children.

Joseph M. Cierniak
European Technical Center, APO

When I first heard about this, my reaction was, "Hard core, man. Hard core! " Of course, the causes of suicide are more often than not unfathomable, but Stereophile's *priorities are not. They commemorate the tragedy of these unfortunate designers of tweaky styli, but I don't remember anything in their pages about the equally tragic decease of the brilliant Deane Jensen, whose microphone and phono transformers were almost certainly the finest in the world, whose circuit analysis program was a remarkable pioneering effort, and whose JE-990 discrete op amp circuit broke new ground in ultralow-distortion amplification.*

—Ed.

The Audio Critic:

...It is obvious that you are doing something right. That is of course that you are demanding that high-end manufacturers stand on the realities of physics....

On the matter of speaker cables, your argument that there are audible differences *but* that these differences are nothing but the results of the interaction of the amplifier, the RLC of the cable, and the speaker is persuasive. It is incumbent on those who believe in the superiority of specific cables, and that this superiority is inherent in the construction of same, to prove it with physical evidence. I should note that Frank Van Alstine has made essentially the same argument in his newsletter. I look forward to your remarks on interconnects and hope that you will comment further on the question of long interconnects and short speaker cables or vice versa.

With regard to the question whether all electronics sound the same as long as they meet your test conditions, I am not so convinced. Circuitry certainly must transform the signal in different ways in different preamps, for example. Certainly I could identify my Conrad-Johnson PV5 as compared to the transistor preamp I had been using. Further, how do you explain the celebrated Carver challenge to replicate the sound of any amplifier selected by J. G. Holt if in fact there are no differences between state-of-the-art amplifiers playing within their design parameters? I should add that I certainly agree with you that the differences between electronics are not as different as "golden ears" would like us to believe....

Sincerely yours,
Harold Goldman
New York, NY

You obviously have a fair grasp of the basic realities of audio but seem to have an incomplete understanding of what happens within the "circuitry" when an input is processed to become an output. Two preamplifiers or two amplifiers having very different circuitry can still operate on identical inputs to produce identical outputs. In other words, the circuitry can be different but the transfer function (output divided by the input) can still be the same, in which case the outputs will be the same. If your Conrad-Johnson and your previous preamp sound different, it's because their transfer functions aren't identical, not because one uses tubes and the other transistors. For example, their output impedances could be different (in fact, I'm pretty sure they are) and/or one could have more of a high-frequency rolloff than the other, and so forth.

Now, the Carver challenge would indeed be rather meaningless if it started out with two amplifiers having the same input impedance, same output impedance, same frequency response, and same gain. Those are the principal parameters that Bob Carver massages to end up with identical transfer functions. In the instance of the J. Gordon Holt challenge, such was certainly not the starting condition, but in some cases it could be, and then Bob would have very little—or possibly nothing—left to do. To that extent, your skepticism is justified.

—Ed

The Audio Critic:

Please consider the two following sets of questions raised by Issue No. 16.

1. The Sound of Amplifiers, Part I

Three key points in the critique of the subjectivist school of audio evaluation are: (a) forgoes all double-blind testing methodology; (b) rejects the oft-repeated findings of double-blind testing; and (c) refuses to play "fair" (pick the Carver controversy of your choice).

In its defense, *Stereophile* has said they will set up and publish a "fair" double-blind test (by their understanding of objectivist methodology). Given the right conditions, they believe their "golden ears" can and will distinguish among amplifiers.

Now, let's make a big assumption. Assume a reputable and methodologically defensible double-blind test is published in *Stereophile* or some other journal. The test shows statistical significance (>95% confidence level) and practical significance (say 13 right out of 16 tries).

Questions: If such a test were published, what is the probability that their results would be accepted by the objectivist camp as being a valid test ? (The conclusion being that either some phenomenon not accounted for by current objective measures is audible or that the audible level of some measured distortion is lower than previous objective tests indicate). More to the point, would the objectivist camp be any more likely than the subjectivist camp to acknowledge they might be wrong?

(I have my own "reconciliation" of the two camps. However, I doubt that you would enjoy wading through my dissertation just to understand my biased view.)

2. The Sound of Amplifiers, Part II

Your basic premise is: All competently designed amplifiers will sound alike if several reasonable conditions are met. Most of your discussion makes reference to amplifiers which behave (more or less) like a voltage source.

Questions: Do any of the reasonable conditions change if you include current-source amplifiers (i.e., the various output-transformerless [OTL] tube designs)? If not, how can a fair comparison be made with an OTL which can swing several hundred volts into a high-impedance load (such as an electrostatic) ?

Do OTL designs affect the frequency response of speakers (particularly electrostatics) in a predictable fashion? If so, does this account in large measure for "subjective" reviewers praising most OTL designs, particularly when used with electrostatics ?

Sincerely,
Barry McClune
Wilmerding, PA

Re your Part I. The subjectivists of the high-end audio press obviously have a political agenda: the $1200 amplifier musn't be allowed to sound as good as the $6000 amplifier, otherwise it's the end of the world. I can't speak for all objectivists, but I and the ones I know well are willing to live with any outcome, as long as it's true. As a matter of fact, I'd be happier if the $6000 amplifier invariably sounded better; it's a terrible downer when it doesn't. So, personally, I'd welcome being scientifically proven wrong in some of these soundalike controversies; a few of my fellow objectivists would possibly have wounded-ego problems—who knows? The point to remember is that truly conclusive objective tests leave no room for argument, whereas assertions of exquisite subjective perceptions always do.

Re your Part II. A highish output impedance makes an amplifier a less-than-perfect voltage source and puts it a small step closer to a current source, but to my knowledge there's no such thing out there at this time as a "current-source amplifier" designed to drive a loudspeaker (it would have to be a very special loudspeaker). What an output impedance of 1.1 ohm can do to the response is illustrated in Figure 8 on page 55 of Issue No. 16. Also, you musn't confuse the two kinds of OTL tube amplifiers that have appeared over the years. One is part of an inseparable amplifier/speaker system, driving a specific electrostatic loudspeaker right off the plates, without an intervening transformer. (Early Beveridge and Acoustat designs come to mind.) The other kind—the only kind that can be A/B'd against conventional amps—is a more or less universal amplifier designed to drive all kinds of speakers. The Futterman OTL amplifier (perpetuated through the mid-1980s by New York Audio Laboratories) was the origin of the species and probably its best example. Inherent design limitations made it quite unhappy with loads below 16 ohms; the original Quad ESL was a very good match to it because of its relatively high impedance. A modern electrostatic like the Quad ESL-63, however, has an impedance characteristic not very different from any number of other speakers, and there's no advantage to driving it with an OTL.

—Ed.

The Audio Critic:

Six months ago I quit smoking and said that if I remained off cigarettes for six months, I'd replace my 10-year-old stereo system (Onkyo receiver, Advent loudspeakers). So for the last six months I've been intently reading the de rigeur periodicals, e.g. yours, *Stereo Review, Audio* and *Stereophile.*

I had read *Stereo Review* and *Audio* (and *High Fidelity)* intermittently over the years, but now I was down to serious business. The ideas I came in with were:

• most high-quality amplifiers would be equivalent;

• FM would be limited by the signal and antenna more than by the tuner;

• CD players would also basically

sound alike.

Even if the above postulates were not precisely true, they were excellent working principles. My main philosophy was that, by a large margin, the speakers had more to do with the sound than any of the components. The listening room and the speakers' placement within it would be the second most important factor. Both of these factors would dwarf the other components' contribution (assuming, of course that they were all of good quality).

Those were my working assumptions. When I started to read *Stereophile,* I was surprised to find that there was a slant or spin to audio that I was unaware of, that of subjectivism. I didn't dismiss this out of hand. After all, oenophiles use the same kinds of words to describe wines. Also, while one reads *Stereophile* there can be a tendency to become enraptured.

I then saw an ad for your magazine in another audio journal, and the premise stated in the ad appealed to me. I subscribed. I liked what I read. I *loved* the debunking. **Eg.:**

- quality amps sound alike when level-matched;
- certain attributes ascribed to preamps are actually determined by the recording process;
- double-blind A/A testing gave differences 35% of the time.

/ love this. It helps to restore some order to the chaos. It affords me armament to do battle with the salesman. It allows me to put my money where my speakers are.

Now to my area of discomfort. Bob Carver.

I know of Bob Carver from his Phase Linear days. He had a well-respected reputation then, and I always felt that he was an innovator. So when I began to read your magazine, I was not at all surprised to find his products highlighted. And there is Carver bashing in the audio stores (e.g., his amps have high wattage *[sic],* etc.). The high-end stores rarely sell his products. (One store here does. I saw the "Amazing Loudspeakers" on two opposite walls facing each other, and when I asked about them the "salesman" said: "They suck.")

Having an extended interview with Bob Carver was fine; reviewing his products is fine; but this last issue is too much: (1) Amazing Loudspeaker review again. (2) Touting the Amazing in the Snell review. (3) Review (I think) for the first time a TV, with a sound system by—who else—Bob Carver.

I don't philosophically or scientifically disagree with Bob Carver. I find the science of your magazine credible up to my level of understanding. But when you praise Carver to this extent, it lowers *your* credibility. And your message in an important one in audio and should not be dismissed.

Dr. Michael Feinstein
Newark, NJ

Your oenophile analogy is a natural one, but check out this quote from the late Frank Schoonmaker's great Encyclopedia of Wine: *"...experts, tasting blind, will rarely vary in their ratings of any given wine, by more than four or five points out of 100. This is a far higher level of unanimity than music critics or art critics or literary critics ever achieve... " It goes on: "...although the terms [wine tasters] use often appear bizarre or pretentious or even ridiculous to those unfamiliar with them, they are certainly more precise than the language of music critics (a 'lyric' tone, a 'warm' voice) or that of painting ('vibrant,' 'sincere,' 'well-organized')." Now, just substitute the word "audio" in the right places...*

As for Bob Carver, he is so much more talented, inventive, and savvy as an audio designer than the high-end cult's typical icons and totems that I enjoy bringing up his name and his products just to see the tweako partisans freak out. Their designer heroes are such crashing mediocrities! Actually, there are only two Carver designs that make Bob a hero in my eyes: the "why didn't somebody think of it before" bass system of his loudspeaker and the incredibly space-efficient power supply of his amplifiers. Those are breakthrough ideas that keep coming up as yardsticks in almost any discussion of speakers and amplifiers. It just so happens, however, that there's very little Carver in this issue.

By the way, another superb designer who isn't politically correct in tweako country is Chris Russell of Bryston. You'll read more about his uncompromising yet highly sensible circuit-design philosophy in upcoming issues. And, again, maybe more repeatedly than some will like.

The Audio Critic:

Gee, Peter, either your hearing or your system is in need of repair. *No* sonic differences between CD players and/or amplifiers if they're current models and evenly matched in sound level?? Come on, you don't really believe that!!

Of course, I used to believe as you *say* you do, but after listening to *truly* high-end (no, *not* Carver) equipment, I became *convinced* that *differences do exist!* In sound-level-matched tests (within *your* stated parameters), my wife has accurately picked differences in CD players, amps, cables, etc., in *true blind tests!*

Spend more time *listening* and less time bashing *Stereophile* and *TAS,* and you'll be happier (so will your readers).

Bob Gash
Lees Summit, MO

Gosh, Bob Gash, if Mrs. Gash can hear these differences that the rest of us can't, we could sure use her in our listening tests. I've always maintained (see Issue No. 16, page 33) that if a single person can provably *hear a difference that hundreds or thousands of others can't, then it's still a genuine difference to which audio professionals must pay serious attention.*

I don't for a moment believe, however, that you really followed my rules; you'd be more specific if you had done so. I bet you matched levels by ear, not by meter within 0.1 dB. No good. I bet you tried each comparison just a few times, not a minimum of 12 and preferably 16 times. No good. I bet you talked to Mrs. Gash while switching back and forth, even if she couldn't see what you were doing. No good.

I'd be very surprised if that wasn't the way it went. You see, my case history is just the reverse of yours; I believed in these audiophile-type differences as recently as five years ago and now I don't. My listening tests as reported in early issues of The Audio Critic were as casual as I think yours are; I set levels by ear and switched back and forth a few times; sometimes I made comparisons sequentially rather than side by side. (At least I always did the necessary bench measurements.) One day, urged by certain fellow practitioners, I matched the levels by meter within 0.1 dB and had the scare of my life. The damned things—I don't remember now whether they were preamps or CD players—sounded exactly the same! I realized that level was the crux of the matter. If you match levels by ear, you'll end up with a mismatch of 0.4 or 0.5 dB at best, and of course you'll hear a small difference, which can then be interpreted as one of "air," "depth," "soundstage," etc., etc., and inflated into a big difference.

As for Stereophile *bashing and* TAS *bashing, where were you and your sharp pen when they started bashing me, long before I retaliated? (If aggressively insisting that 2 + 2 = 4 constitutes retaliation.)*

-Ed.

In Loudspeakers, Is a Good Big One Always Better than a Good Little One?

By Peter Aczel
Editor and Publisher

On the cutting edge of the art, which is what we're investigating here, unexpected things happen. The laws of physics favor the good big one over the good little one, but what about a superb little one?

Once again I must refer new readers of *The Audio Critic* to earlier issues in which my approach to loudspeaker evaluation was explained at length, particularly Nos. 10, 11, 14, and 16. I can't possibly go over the same ground each time I review a new speaker system, even if the review is then not quite self-contained and self-explanatory.

Fried Q/4

Fried Products Company, 7616 City Line Avenue, Philadelphia, PA 19151. Model Q/4 compact 2-way loudspeaker system, $498.00 the pair. Tested samples on loan from manufacturer.

Irving M. (Bud) Fried is one of the founding fathers of consumer audio in America; I first became aware of him in the late 1950s, when he was importing the original Quad ESL from England, thereby rising to high priest status in the eyes of us purists. Later he became a loudspeaker manufacturer, under various brand names, of which Fried Products gained permanence. Under that name his speakers have been reviewed in these pages off and on since 1978, mostly favorably. My overall impression has always been that Bud Fried is both knowledgeable and realistic (i.e., not tweaky) about loudspeakers—although he is (or at least used to be) somewhat reluctant to accept the fact that a transmission-line enclosure obeys the same laws of physics as a vented or closed box—and I know that he is attuned to the sound of live music.

The Fried Q/4 under consideration here is a sleeper. Who would have thought that a rather chintzy-looking pair of bookshelf-size boxes listing for less than $500 would sound better than 90% of all other speakers, regardless of price? What a buy! If I accept, as I must, the size-related limitations of the *Q/A*—lack of window-rattling deep bass, lack of very high SPL capability, same driver for bass and midrange—then I honestly can't think of anything I'd want to change in its design at this price except the outside of that severely "entry-level" box, which has no provisions to minimize diffraction. Everything else appears to be optimal.

The 8" plastic-cone woofer and 1" cloth-dome tweeter are American-made (by United Speaker Systems in Florida) to Fried's specifications; the tweeter faceplate has the Fried logo engraved on it. The crossover network, which appears to be third-order (18 dB/oct slope) with the tweeter polarity reversed, was computer-designed by Ken Hecht of USS. The enclosure features what Fried calls a "line tunnel," a kind of shrunken transmission line exhausting into a stuffed-up slot; to me it looks as if it had much the same effect as "aperiodic loading." The cabinet is quite solidly constructed, but the finish is cheap-looking vinyl. You can see that the money went into the innards.

The dome tweeter is very impressive; it goes out flat to 30 kHz with only the slightest shelving starting at 13 or 14 kHz but still staying above the -3 dB line. That profile, in combination with the absence of even the smallest peaks (other than those due to cabinet diffraction), results in smooth-as-silk string tone on classical music and totally nonfatiguing highs on any kind of program material. I think this tweeter is in the same league with the JBL pure-titanium 1" dome; I wish I could have tested them side by side, but I no longer had the JBLs on the premises. The Fried tweeter is crossed over at 3 kHz.

The 8" woofer is also excellent—it *has* to be when crossed over as high as 3 kHz. I could discern no bad behavior in the crossover region. The bass response profile of the system is essentially that of a well-damped closed box, with a 12 dB/oct rolloff. The -3 dB point is at 60 Hz, but the gradual rolloff allows strong fundamentals well below that frequency. Indeed, the bass of the Q/4 is quite remarkable for such a small box, so that many users will feel no need for a subwoofer. On the other hand, the specification in the literature claiming ±3 dB response down to 37 Hz is absurd. The fundamental resonance of the system, as indicated by its impedance peak, is around 74 Hz.

A full frequency sweep of the Q/4 shows an ever-so-slight elevation of the low-frequency range as compared with the midrange and elicits a few low-level buzzes from the cabinet. The overall response is quite flat, with just a mild rolloff of the highs off axis. Tone bursts reveal no storage to speak of; square pulses are more or less recognizable with "sweet-spot" microphone placement but show a negative-going preshoot (reverse-polarity tweeter) and a very chewed-up top (to be expected with the given crossover network). I find no serious fault in any of these results.

The sound of the Fried Q/4 is, as I said, outstandingly good—smooth, transparent, uncolored, and highly defined. The ultimate spatial detail is missing, most probably as a result of diffraction due to the sharp edges and corners of the cabinet (Audio Concepts' cabinet design, for example, is better for imaging); even so, I can't think of any speaker system under $500 the pair that can equal the Q/4, while I can think of any number of $1000 to $2000 speakers that sound a lot less accurate and musically satisfying. You can't go wrong with this "good little one."

Snell Type B

Snell Acoustics, Inc., 143 Essex Street, Haverhill, MA 01832. Type B floor-standing 4-way loudspeaker system, $4200 the pair. Tested samples on loan from manufacturer.

This is a big one, *the* big one designer Kevin Voecks has been working on and talking about for years. It's over four feet high and as wide as an NFL linebacker. How good is it? Very good indeed, but not as good as some of the wildly enthusiastic early reports may have lead you to believe. In my opinion the Type C/IV, at half the price, is in a number of ways a better design. Now, I don't want anyone to walk away with the impression that ***The Audio Critic*** gave the Type B "a bad review," so please read my comments very carefully.

First of all, let's all agree that no unheard-of miracles should be expected from a forward-firing box speaker using conventional dynamic drivers, no matter how well-engineered it is. (All right, the enclosure of the Type B isn't entirely box-like—it's a pentagonal column—and there's the rearward-firing extra tweeter which is the Snell hallmark, but the generic classification still holds.) Not even Kevin Voecks can come up with a totally new and vastly more accurate sound within that format. The B has one more woofer and one more midrange driver than the C/IV but it isn't a startlingly different design and therefore it won't take you into a startlingly different world of listening. That's just common sense.

The driver complement of the Type B consists of two 10" woofers, each in its own sealed cavity (and one of them not really a woofer but a weird sort of bump-up filler—more about that in a moment); two 5" midrange drivers and a 1" dome tweeter in the so-called D'Appolito arrangement (mid/tweet/mid in a vertical line); plus the rearward-firing 1" dome. The woofers are mass-loaded at the apex (obviously because off-the-shelf units having the desired cone mass were unavailable, not because it's a high-tech feature), and the two of them are actually *crossed over* to each other at—believe it or not—40 Hz, with 12 dB/oct slopes. Above 40 Hz, the bass is handled by the front-facing woofer; the other woofer is aimed away from the listening area and is rolled off above 40 Hz with a humongous LC combination—but it also rolls off naturally in closed-box fashion below the mid-30s, so it produces only a filler bump! This is supposed to eliminate certain interaction problems at the speaker/room boundary; I have no opinion on that at the present time. The upper-woofer-to-midrange crossover slopes are 24 dB per octave; the D'Appolito cluster uses the prescribed 18 dB per octave slopes; the rear tweeter just has a capacitor in series to roll off frequencies below 5 kHz. All drivers are wired in phase. I'll call it an unconventionally configured conventional dynamic speaker system.

On the basis of nearfield measurements (the Don Keele method that so neatly tracks the anechoic curve), I'd say that the bass enclosure of the Type B is a 36 or 37 Hz box—or, rather, double box. The C/IV has a much lower bass cutoff (-3 dB point), but of course the B can handle more low-frequency power and its rolloff is more gradual. On music with lots of bass, particularly timpani, bass drum, heavily bowed double basses, etc., the B is exceptionally potent, clean, and well-controlled. The bottommost bottom, however, isn't in evidence—it's no Velodyne. For $4200, I want it all, from 20 Hz on up, and I'm not getting it. The Snell specification of ±1.5 dB from 20 Hz to 20 kHz "in anechoic half-space with 1/5 octave averaging on the listening axis" is a computer-massaged conversion from full-space reality and not particularly meaningful at low frequencies.

From the bass frequencies on up I found the Type B to be extremely flat on axis up to 20 kHz and beyond; off axis the response is still very flat up to 15 kHz. Truly excellent. The two midrange drivers come in at 275 Hz, the aluminum-dome tweeter at 2.7 kHz. The latter is identical to that in the C/IV, with the same peak at 25 kHz—inaudible, of course, and therefore of no consequence. This same Vifa unit is also used as the rear tweeter of the B, instead of the cheap-but-good Audax that Snell puts into all other models. The rear-panel control for the front tweeter affects only the level matching to the midrange, not the contour of the treble response. The rear tweeter has only an on/off switch. Overall, I'd say that the performance of the B in the frequency domain is impeccable.

In the time domain, my square-pulse test proved once again that 4-way speakers with steep crossovers can have no coherence whatsoever, but then Snell has never had an interest in coherence, and there's authoritative support in the literature for that point of view. Somewhat more disturbing was the tone-burst test, which showed quite a bit of spurious energy between the tone-burst envelopes in the midrange and the mid/tweet crossover range. This may have been due to interference patterns instead of storage; sometimes it's hard to tell the difference. Suspecting the midrange drivers, I then discovered that they have hardly any piston excursion but operate almost entirely in the transmission mode. That requires very good termination (i.e., dissipation of standing waves), and I've never seen it done 100% right. These drivers, also made by Vifa, are being used here for the first time by Snell, apparently because of their very flat response on and off axis. I'm not convinced, however, that they introduce no coloration under certain signal conditions nor that they can handle the most taxing peaks on vocal music.

I find something vaguely "not right" in the sound of the Type B that I'm inclined to attribute to the midrange drivers, especially in view of the less-than-perfect tone-burst response. It's a subtle coloration or lack of ease or stuffed-up quality, hard to describe and so slight that many will deny it. Another possible source of it is the cabinet, which responds with a distinct pitch when struck in certain places with a small padded hammer I use for the purpose. It's very difficult to build a totally dead *large* cabinet without going to extremes in the manner of Avalon Acoustics. The cabinet pitch is also approximately in the range that gives me discomfort.

None of this should be interpreted to mean that the Snell Type B doesn't sound good. Of course it sounds good! It's a big, authoritative, dead-flat, clean, obviously high-end speaker. But the absence of the deepest bass and the slight flaws just mentioned make it less than the "ultimate" conventional speaker system, which is what I expected from Snell. It seems to me that the relatively simple C/IV format is easier to implement than the considerably more complex Type B architecture. It's possible that no one could have done it better, with off-the-shelf drivers and just a normally well-built cabinet, to retail for $2100 per side. My respect for Snell is certainly not diminished.

Win SM-10

Win Research Group, Inc., 7320 Hollister Avenue, Goleta, CA 93117. SM-10 Broadcast Monitor (2-way coaxial loudspeaker system), $6250.00 the pair, including stands. Tested samples on loan from manufacturer.

Dr. Sao Zaw Win, as faithful readers of *The Audio Critic* know, is the Cambridge-educated Burmese-American scientist/technologist who is equally at home in a radiation-proof life-support suit cleaning up some unsophisticated nuclear mess and in an electronics laboratory designing high-end goodies for us audiophiles. (Yes, he prefers the latter.) I'll say one thing about his work: he never gets involved in anything unimportant. You won't catch him putting the finishing touches to the industry's 927th tube preamplifier. His turntable of 1978, his FET phono cartridge of 1987, although not timely enough for commercial success, were landmark designs; his new loudspeaker is that and more: a classic that promises to be the standard for small monitors for years to come. The SM-10 represents a total concept, one that started with a blank sheet of paper; it isn't a repackaging of old ideas into yet another expensive new toy for the insatiable high-end consumer.

Let me say right up front that, for applications where the deepest bass and the widest possible dynamic range are of less than the highest priority, this is the finest loudspeaker known to me. It's simply a cleaner window than any other for admitting the sound into the room. The price is brutally high, but if one disregards the retailer's $2000 markup)—inevitable under the high-end audio industry's current distribution system—the money is right there in the speaker box, in terms of both hardware and development work.

The SM-10 is a small speaker—19¼" high, 12¼" wide, 10½" deep—that mates to a dedicated metal stand for seated ear-level elevation. The rectangular box, made of 1¼" thick Medex, has perfectly rounded edges, all 12 of them, and is finished with coat after coat of special Italian black lacquer until it gleams like Napoleon's sarcophagus. I understand that the cost of a pair of finished boxes to the manufacturer is over $800. Sheer insanity, but very beautiful. The geometry and construction of the box are the result of extensive Finite Element Analysis, which is beyond my ken, but I can report that the box is deader than any other I've ever tested, totally unresponsive to my knuckles or my little padded hammer.

The *raison d'etre* of the Win SM-10 isn't the construction quality, however; it's the extraordinary 2-way coaxial transducer, which is radically different from anything used in any other design. Both woofer and tweeter have completely flat diaphragms, and their deployment is both coaxial and coplanar, in other words like a small circle inside a fat ring in the same plane. The size of the woofer is roughly equivalent to that of a conventional 8" unit; the tweeter would be described as a 1" dome if it were a dome and not flat. The crossover frequency is 3.2 kHz.

The woofer diaphragm is made of thin layers of woven carbon fiber compressed with silica gel; the tweeter diaphragm is made of compressed mica and alumina. Not exactly your everyday cone/dome materials. (Sao Win is a specialist in physical chemistry and the testing of materials, so it's no surprise that he shows some originality in this area.) Finite Element Analysis was also used in the design of the diaphragms and other physical components of the drivers; the magnet and voice coil designs are based on computer programs (Poisson/Superfish group of codes) developed at the Los Alamos National Laboratory by Ron Holsinger, who was once introduced to me as "Captain Magnet." Indeed, there's no seat-of-the-pants design evident anywhere in this speaker; its R & D credentials as documented in the very detailed technical literature that comes with it read like those of a major government defense project.

I was particularly impressed with the optimization of the tube-vented enclosure. The size/bass/efficiency trade-off in this system is probably the smartest I've ever seen; it was done with the readily available LEAP software, I'm told. The -3 dB point on the response curve I took was at approximately 44 Hz, maybe even a hair lower; the lowest 0 dB point was at 50 Hz, and the bass response was dead flat from there on up. (For those who care about such things, the box is tuned to 56 Hz, and maximum output from the rearward-directed vent is at 60 Hz. These figures could be off by a hertz or so.) When I first played the speakers, before taking any measurements, I suspected a little bit of sophisticated cheating in the bass—a bump of a few dB in just the right place—because the bottom end didn't appear to be missing at all, but later I realized that Sao Win is too much of a purist for that. No, he manages to give you the flat bass extension of typically much bigger boxes, and still with fairly high efficiency (88 to 89 dB). That's close to the ragged edge of the laws of physics.

Taking the overall frequency response curve of the Win SM-10 is quite a bit easier than in the case of a speaker system with a woofer here, a midrange there, a tweeter over there—and who knows where they coalesce? You just point the measuring microphone at the bull's-eye of the coaxial assembly, on and off axis, and all you have to worry about is not picking up room reflections—everything else is quite uncritical because the wave launch is symmetrical over the entire frontal hemisphere. Piece of cake. I can therefore confirm with some degree of certainty the claimed response of ±2 dB from 55 Hz to 20 kHz on axis; indeed, it's better than that except for a little blip or wrinkle in the crossover region, which takes up the full ±2 dB tolerance. (More about that in a moment.) Although the specs don't says so, the flat axial response continues out to 30 kHz. Off axis the tweeter response is still very flat, but there's a dip in the crossover region that becomes quite marked as the measuring angle is increased.

It would seem, in light of the above, that the crossover network isn't quite as fanatically optimized as the rest of the speaker. I don't want to make a federal case out of this because the measured response is still so excellent and the audible results superb, but I have a feeling that the next production run (the first, from which my test samples came, is sold out) will sound even a little better because it will incorporate a slightly reworked network. (Yes, retrofits will be available, the network being outside the speaker, attached to the stand.) The basic concept of the crossover is rather similar to the one developed by John Bau for his Spica speakers. The woofer is rolled off with a fourth-order Bessel lowpass filter; the highpass filter for the tweeter is mathematically derived to fit the lowpass section as closely as possible in terms of amplitude matching and the desired constant group delay characteristic. My theory—purely conjectural—is that the wave launch from a ring-shaped radiator such as the SM-10's woofer/midrange driver (as distinct from a circular piston) wasn't part of the mathematical model used in the optimization. I could be totally off the wall here; what Sao Win told me was that the crossover frequency will most probably be moved down from 3.2 kHz to 2.7 kHz—for which the tweeter has ample bottom-end room—and the response in the crossover region will then be expected to flatten out considerably. It's a minor problem in any case.

My time-domain tests painted a highly satisfactory picture. The woofer and tweeter diaphragms both move forward in response to a positive-going pulse. A square pulse input produces a highly recognizable square pulse output, although with some imperfections; the top of the pulse shows a leading-edge spike followed by lots of wrinkles, indicating that the crossover doesn't quite allow perfect coherence. That could change, as I said, in the next run. The spaces between the pulses are very clean. Tone bursts also indicated that there is indeed some kind of mild crossover glitch but revealed no energy storage in the diaphragms. The high-tech "dead" materials are doing their job.

If you've been waiting for a pornographically explicit description of the sound of the Win SM-10, I'll have to disappoint you. It simply reproduces the music. The reproduction is so transparent, uncolored, and clean that the sound just *is*—it isn't *like* this or *like* that. "Exquisite" was one comment by a casual listener. I don't know of a speaker at any price that equals the SM-10 in this respect. Deeper bass, bigger whacks on the Telarc bass drums, more spectacular orchestral climaxes, a more room-filling sound I've heard. Greater accuracy and greater beauty on beautifully recorded material I haven't. Not that the SM-10 is a wimp. It gives you a pretty ballsy sound when driven hard with a big amplifier—not to worry, it can take it—and it presents a surprisingly big soundstage in an equilateral triangle setup. I can't imagine a better speaker for a recording engineer or a record producer to take on the road in his car, or for a well-heeled audiophile to listen to in a city apartment. Eventually, I'm told, there will be a Win subwoofer; meanwhile the SM-10 is merely the bass champion of minimonitors.

I'm inclined to think that the main reasons for the superb sound of the SM-10 are the perfectly symmetrical, diffractionless wave launch from a virtual point source and the extreme precision of construction. Remember, even the Quad ESL-63, which is also based on the point-source concept, has limited horizontal dispersion compared with the vertical. Another difference is that the ESL-63 is bumped up at 50 Hz, whereas the SM-10 isn't. Even so, if you like the Quad, you'll love the Win. They come from the same school of wave-front modeling.

Other comparisons that should be made are with the Wilson WATT, comparably priced at retail but with far inferior hardware inside the gorgeous cabinet, and the KEF and Tannoy coaxial designs, which are also basically point-source radiators but have the serious disadvantage of firing the tweeter through the "megaphone" formed by a conventional woofer cone—unlike the flat, flush, and coloration-proof SM-10 transducer. The Win wins on all counts. It's the good little one that can beat a good big one. •

Subwoofer (late addendum)

Audio Concepts Sub 1

Audio Concepts, Inc., 901 South 4th Street, La Crosse, WI54601. Sub 1 "Synthesized Bandpass" subwoofer, $749.00 the pair (direct from Audio Concepts, fully assembled, including shipping charges). Full kit, $649.00 the pair (including shipping charges). Tested samples on loan from manufacturer.

This very interesting subwoofer arrived too late for a really thorough evaluation, but it deserves to be included here just to make sure that our readers are aware of it. It fills a genuine need by providing deep, clean bass in small and medium-sized rooms at a ridiculously low price in a very compact package. That's not something to be sneezed at. The design must be judged on its own terms, however, not in competition with something like the Velodyne ULD-15 (reviewed in the last issue), which costs almost five times as much. In the right environment, a pair of Sub l's in combination with the Audio Concepts Sapphire II minimonitors will give you superior performance even by high-end standards, and the bill will be only $1538.00 for the four-piece system. (No, a single Sub 1 won't do it; it isn't designed for L+R matrixed operation.)

The Sub 1 is just a little over two feet high and slightly over a foot in both width and depth. Not at all big for a serious subwoofer. The 12" dual-voice-coil driver faces downward, firing into the floor through a space about two inches high and open on all sides. Needless to say, no highs emerge that way. I suppose that's what makes the Sub 1 a "Synthesized Bandpass" subwoofer because the built-in crossover network has a fairly conventional second-order lowpass section. The highpass section is just a capacitance in series with the main speaker; the crossover frequency could be said to be anywhere from 80 to 100 Hz, depending on how you define it when the transition is so gradual. The box has little stuffed-up holes in the bottom panel around the woofer to give the system the "aperiodic" bottom-end characteristic favored by Audio Concepts. It's all pretty simple but it works (in the right environment, I must add again).

In the June 1991 issue ("Three/91") of *Speaker Builder* Gary Galo reviewed the kit version of the then brand-new Sub 1. Although I suspect that Gary Galo's audio philosophy departs significantly from mine, I find nothing to contradict in his very thorough review after having made my own tests, which as I said were somewhat hurried. The nearfield response I obtained is exactly the same as he shows: just a peak at 40 Hz with an immediate rolloff below it and above it. That's with the cone firing into open air, obviously not representative of the woofer's principle of operation. The nearfield response taken at floor level, sticking the microphone into the two-inch airspace below the cone when the Sub 1 is standing correctly, is not shown by Gary Galo; I found that the curve was very similar but indicating 3 dB higher deep-bass efficiency, the rolloff below 40 Hz being parallel to that of the open-air curve, 3 dB above the latter. Clearly, then, that's not the way the Sub 1 works, either, although the loading effect of the floor is part of the idea. It appears that the design depends very heavily on the low-frequency boost inherent in the boundaries of the room, and a not very large room at that.

Gary Galo shows an in-room farfield response of ±0.5 dB from 22.5 to 90 Hz, with no satellites connected. To do this measurement correctly takes more time than I had, but I'm quite willing to believe that the response is as good as that—in *his* room. In my room (approximately 22' by 20' by 9'), I had a lot of trouble hearing the bottommost bottom, just about regardless of placement. Finally, using the Bill Rasnake technique (see Issue No. 13, pp. 43-52), I found two impractical locations for the two boxes that worked very nicely—the bass drums, double basses, and organ pedals sounded just fine. I say impractical because the Sub 1 units were so deep into the corners that the Sapphire II's I was testing them with had to be set up much closer together and much further into the room, creating minor bass-to-lower-midrange transition problems. A crossover close to 100 Hz isn't really low enough to make the subwoofer positions totally uncritical; with 60 to 70 Hz the separation from the satellites would have made no audible difference. Using that nicely woofing but impractical setup I heard lower-midrange colorations that weren't there with Sapphire II's used full range. (See David Rich's caveats regarding the woofer-to-minimonitor crossover situation in the last issue.) In a smaller room, of course, there would have been less separation regardless of the subwoofer locations.

The bass I heard with this not entirely satisfactory deployment was notably clean and well-controlled, without a trace of hangover, confirming the highly damped (low-Q) response claimed for aperiodic loading, which my measurements also showed. My overall conclusion has to be that such a generally good impression under unfavorable conditions implies outstanding results in smaller rooms, where the room boost is always greater and the probable separation between a well-placed subwoofer and its satellite smaller.

Erratum:

In my review of the Carver "Amazing Loudspeaker" Platinum Mark IV (Issue No. 16, pp. 12-14), the price of the loudspeaker is quoted several times as $2199.00 the pair. That was true when I last looked before writing the review—and only for the less costly oak finish—but by the time the issue was published it was incorrect. Price revisions are a continuing process at Carver Corporation, so even if I had quoted the price correctly in the review it would no longer be correct by now. Here are the correct prices as of January 1, 1992: in natural oiled oak veneer, $2499.95 the pair; in piano-lacquer black finish, $2899.95 the pair. I much prefer the black lacquer finish—it looks appropriately High End. As for the new prices when judged against the quality of the product, they're now merely astonishing instead of being totally unbelievable.

How to Squeeze Low Bass from Small Boxes: a Survey of Techniques and Trade-Offs

By Christopher Ambrosini, Ph.D.

This inquiry into the feasibility of generating long wavelengths from low-volume transducers attempts to bring a modicum of structured understanding to a subject that has long suffered from vagueness, pseudoscience, and conflicting claims in the popular audio press.

Editor's Note: *Christopher Ambrosini is the* nom de plume *of a highly accredited audio journalist and techie-of-all-trades, who for reasons that have absolutely nothing to do with* ***The Audio Critic*** *doesn't wish to sign this article with his real name. The Ph.D., however, is real, although his academic background is not in electroacoustics.*

* * *

Since much, much balderdash has been written about subwoofers both in manufacturers' literature and in reviewers' assessments, the topic of subwoofers seemed to cry out for clarifying remarks from ***The Audio Critic,*** particularly in regard to the practicality of making a subwoofer of relatively small dimensions—let us say less than four cubic feet. What follows is an attempt to explore the basic engineering issues and examine some of the more intelligent design approaches.

But before I delve into the minutiae of subwoofer design, I'd like to say just a few words about the place of this bastard product category in a high-performance music system.

Seldom stated fact: In adding a subwoofer, one is essentially constructing an entirely new speaker system, and unless the subwoofer has been specifically designed to complement a specific wideband speaker system of limited bass output, the integration of the subwoofer into the audio system will present the consumer with formidable problems. The consumer will be faced with the task of determining the crossover point and configuration, arriving at appropriate wave-launch characteristics for the system and addressing problems of room geometry which may not have been apparent with unaugmented bass. In aggregate, these problems and their solutions are worth at least an article in themselves, and should be recognized by anyone contemplating the purchase or construction of a subwoofer. I haven't space to discuss these problems here, but rest assured, they are considerable.

But if subs as a subspecies are highly problematical, they are not without utility. A properly designed, properly integrated subwoofer can significantly—and I think audibly—reduce intermodulation distortion in many, perhaps most, wide-range speaker systems and of course can add low bass extension and impact to all but the largest systems. And let us not ignore the fact that subs are immediately impressive—something you can show off to those friends of yours unmoved by "air" or "liquidity."

Unfortunately most subs that really deliver the bottommost notes are rather enormous, and all but a tiny handful of the exceptions are dauntingly inefficient or dynamically limited. And this is precisely the point I wish to address in this article.

Is then an everyman's subwoofer even possible? A sub that goes really deep—say below 25 Hz—yet is small, accurate, dynamic, and efficient ?

Many manufacturers will be quick to assure you that such desiderata are fully obtainable in their products, but as snake oil is the common currency of our industry, we need not take such claims at face value. If there's one thing that should absolutely be hammered into the minds of consumers regarding the subject of bass reproduction, it's this: downsizing the woofer enclosure will either lower the efficiency or raise the bass cutoff frequency—one or the other, no exceptions, no mercy. Compactness, efficiency, and bass response are an "eternal triangle"—each of them profits at the expense of one, or both, of the other two. Let us therefore consider the range of realistic possibilities dispassionately and comprehensively.

Defining the category and the problem.

A subwoofer is nothing more than a loudspeaker of specialized function—one designed to reproduce frequencies no higher than 200 Hz and sometimes as low as 15 Hz. Like any high-fidelity loudspeaker, a subwoofer is intended to maintain fairly level frequency response through its passband.

Now in order for any cone loudspeaker, subwoofer or no, to maintain level frequency response with descending frequency, it must maintain constant acceleration through the passband. An inevitable consequence of maintaining constant acceleration is a quadrupling of driver displacement with each descending octave. Said displacement requirement, while rather enormous on the face of it, is essentially inconsequential at frequencies above 200 Hz. A small cone moving mere fractions of a millimeter in the magnetic gap can easily produce a 100 dB output above 200 Hz, and quadrupling that small displacement an octave down poses no problem. But adding a further fourfold increase an octave further down is another matter, and multiplying by four again to reach down to 25 Hz obviously involves a very large increase in displacement—from millimeters to the order of a centimeter. And that increase takes all but a very few drivers to the limits of their excursion and beyond.

Something to keep in mind: a one centimeter peak-to-peak excursion is very considerable for a woofer. Very few commercially available drivers will do more than that, and of those that do, most become highly nonlinear as they approach the limits of their excursion because the voice coil is interacting with fewer lines of magnetic flux than at lower excursions. (The highly specialized, dedicated 12-inch woofers of the Carver "Amazing Loudspeaker" are among the rare exceptions, with three centimeters linear travel and five centimeters with some nonlinearity.) Flux density may be linearized over long excursions by the use of long voice coils or design stratagems such as the use of magnetic shorting rings, but optimizing motor design for long excursion tends to be very expensive, and most manufacturers opt for another solution—using bigger cones.

A large-diameter cone will always displace more air than a small-diameter cone for a given excursion, and thus it will not have to move as far at lower frequencies as its smaller brethren. Opting for big cones in subwoofer applications means, however, that the designer is faced with larger enclosure requirements because of the need for a greater panel surface area to mount the driver and the generally higher box-volume requirements entailed by the use of bigger drivers.

In other words, the driver itself poses a major obstacle in achieving significant size reduction in subwoofers. There just aren't that many small drivers of reasonable cost that perform satisfactorily at low frequencies, and for realistic subwoofer applications a ten-inch cone diameter represents a practical minimum; in fact, very few systems using tens can produce any appreciable acoustic power below 30 Hz. Twelves and fifteens are generally much more appropriate, and eighteens are not overkill.

But if the driver itself constitutes one fairly hard limitation in terms of downsizing, the enclosure is apt to impose far more intractable limitations—in other words, with most enclosure designs, you'll never reach the point where the box is too small to accommodate the woofer.

The box conundrum.

Now let's examine why enclosures inevitably impose considerable interior volume demands for subbass reproduction—though of course not all designs are equal in this regard.

It's a well-known fact that in attempting to use a woofer in free air or on a flat baffle, one experiences falling output in the bass region as a result of the cancellation of the primary output by the rear wave. The cancellation will begin to occur when the half-wavelength being reproduced equals the shortest dimension of the open baffle. While open baffle speakers capable of low bass have been made—the Carver "Amazing Loudspeaker," the Enigma subwoofer, and the Celestion System 6000 come to mind—the approach has its limitations, requiring as it does a very large baffle area and/or active equalization, plus highly specialized, dedicated drivers. Hardly the compact, unobtrusive, everyman's subwoofer which we are contemplating.

If we eliminate the flat baffle approach, then we're forced to use a box. Now all boxes under the sun can be placed in two great kingdoms: the kingdom of boxes where only the primary output of the driver is permitted to reach the listener, and the kingdom of boxes with resonant cavities, which either supplement the primary output from the front of the driver or else constitute the whole output themselves. Within these two categories are many variants, but no enclosure design falls outside of this simple division, and within each kingdom are not one but several design variants characterized by remarkably low volume per a given degree of bass extension. I shall describe a number of these variants, but first I must say something about the acoustical behavior of air in enclosed spaces.

Whatever the enclosure design you choose to consider, it encloses a volume of air having the mechanical properties of mass, stiffness (the inverse of compliance), and acoustical resistance or friction. These mechanical properties may be considered to be analogues of inductance, capacitance, and electrical resistance in a circuit, and like the latter they may be manipulated within an acoustical circuit to create a tuned resonance. In a loudspeaker enclosure designed to load electrodynamic cone drivers, the reactive properties of mass and stiffness are always dominant, and it is fairly obvious that the values of both are dependent on the volume of air in the enclosure. A large volume of air is relatively compliant and lacking in stiffness, while at the same time it has considerable mass. Reduce volume and mass is reduced as well, while at the same time compliance drops.

And therein lies the problem for the subwoofer maker. The air in the enclosure is in series with the acoustical circuit of the driver, and together they form a single tuned acoustical circuit, or, more correctly, a series of circuits because all loudspeaker systems have multiple resonances. A driver loaded into a small volume reacts to the stiffness of the air in that volume and resonates at a higher frequency than it would in a large volume. The driver also sees the mass of the air behind it, and if the enclosure happens to be vented, there will be three resonant frequencies in the system due to three different mass/compliance relationships inherent in that format. Naturally, a lower mass in each case will resonate at a higher frequency than a larger mass if compliance is constant.

So now the problem becomes clearer. Small volumes of air tend to resonate at high frequencies. Try to load a woofer with a low free-air resonant frequency into a cavity with a high resonant frequency and something untoward is going to happen. In the case of a simple sealed-box design, a too small enclosure volume will simply give you a higher cutoff frequency and a somewhat peaky bass response. In a vented system, an undersized box will to produce two uncontrolled resonances with consequent boom and overhang, and a high bass cutoff as well. In neither case will you end up with a subwoofer because bass response will be sharply abridged. Thus any crude, brute-force attempt to make a subwoofer by simply cramming a big woofer into a little box is going to result in a botch.

Now we get to the meat of the matter—how precisely does one slither around all these troublesome acoustical plain facts and get a small tuned cavity to behave like a big one? Or is it even possible?

Well, yes, it is possible. There are quite a number of ways to make a small cavity behave as if it resonated at a lower frequency in order to permit the construction of a smallish subwoofer. All of them involve penalties in other areas of performance, however.

The trade-offs involve efficiency, maximum SPL, bandwidth, and transient response. In each design solution the trade-offs are different, so it's best to consider them each in turn.

Perhaps the best way to examine the various downsizing design approaches is in ascending order of engineering difficulty. By that measure we would start with the acoustic suspension speaker and end with the electronically assisted systems such as motional feedback woofers and the ACE-Bass.

And so we begin.

The acoustic suspension speaker.

I won't devote a lot of attention to this time-tested and now mundane design approach, which represents the first commercial attempt to produce a modestly sized enclosure that would support deep bass extension. The principle at work here is simple. Simply stated, the driver and the enclosure are considered together as a single acoustic circuit. The enclosure is completely sealed (see Figure 1), and when the driver is loaded into this "air spring" enclosure it sees only the stiffness of the air, not its mass reactance, and thus the driver is tuned to a higher frequency than its free-air resonance by that stiffness component. However—and this is crucial—the elevated frequency can still be very low, in spite of the small cavity, if the design parameters are correctly chosen.

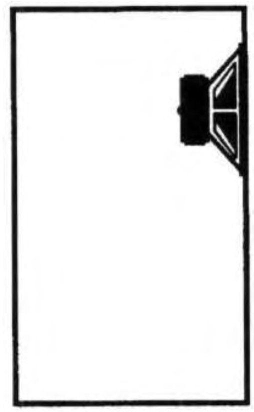


Figure 1: Basic configuration of the acoustic suspension speaker enclosure.

The basic design innovation inherent in the acoustic suspension system was to take a relatively small-diameter driver—usually anywhere from an eight-incher to a twelve—and to tune it to a very low frequency by making the cone structure both massive and highly compliant. (The cone would also be designed for relatively long excursion to make up for the lack of piston area.) Special woofers had to be developed to permit practical acoustic suspension designs because the typical cone of the early fifties (when the design first appeared) tended to be extremely stiff and uncompliant. But once this design objective had been met, the implementation was easy.

Acoustic suspension systems tend to yield most satisfactory bass response when designed to have a Q of 0.7, which represents a maximally flat Butterworth filter response. For this reason it is a little difficult to compare them with classic infinite-baffle designs, such as the Bozak Concert Grand, which preceded them and which were tuned to a Q of 0.5 or thereabouts. Because of this low-Q tuning, the latter had qualitatively different bass response with excellent damping but with a 6 dB rolloff beginning at a relatively high frequency. Disregarding this quality issue for a moment, we will observe that a Bozak Concert Grand, which had approximately twenty cubic feet of internal volume, had a 3 dB down point of 28 Hz. Modern acoustic suspension designs achieve similar bass extension with one fifth that volume.

Obviously the compact air-suspension speaker, under the aegis of Acoustic Research, routed the lumbering old infinite baffles pretty quickly. After all, why give up half your living room and pay for all that cabinetry when a smart little inexpensive box will give you honest deep bass?

Why indeed, except that the massive cone required by the acoustic suspension design gave up a good deal in efficiency, and the dimensions of the cabinet itself forestalled the use of the Concert Grand's heavy artillery—a full four 12" woofers per side. No, the ARs did not outperform the Concert Grand as bass reproducers. The downsizing was achieved at a substantial penalty, as it nearly always is.

Still, the acoustic suspension system was an ingenious if simple idea, and its advocates eventually came up with an ingenious and simple modification that permitted further downsizing of the cabinet, and at no further penalty in efficiency!

Designers found that by stuffing the inside of the box with fibrous damping—such materials as wool, fiberglass, kapok, and Dacron have been employed—the enclosure behaves as if its interior volume had been increased by roughly fifteen percent. (Theoretically even higher equivalent volume increases can be achieved by this means.) The designers here were exploiting the isothermal principle of operation. Here's how it works:

When air is compressed, it gets warmer. Example—a bicycle pump gets warm, even hot, when it is used. The opposite cools air. Example—when air is rapidly let out of a tire, the tire and valve stem get cold. In an unfilled cabinet, when the woofer moves backward into the enclosure, the air is compressed and normally gets warmer. The warmer air, trying to expand, pushes on the woofer, thereby increasing the stiffness that the woofer "sees." When the woofer moves outward, the air is rarified inside the cabinet, cools down, and "pulls" the woofer back. This net pushing and pulling which the air imparts on the woofer helps set the low-frequency system resonance. When stuffing material is put in the cabinet, it acts like a heat sink because its long fibers are in intimate contact with the air molecules in the cabinet. The stuffing absorbs heat when the air is compressed, keeping the air cool so it doesn't push back as hard, and releases its absorbed heat into the air when the woofer moves outward, preventing the air from cooling as much, and thereby doesn't "pull" the woofer back into the cabinet as much. The net effect is that the woofer "sees" an air spring that is more compliant, the same as if the box were larger by approximately fifteen percent. Today virtually all acoustic suspension speakers with any pretensions to quality are densely stuffed in this manner and can therefore be made smaller than they would have to be without the stuffing.

All in all, the stuffed acoustic suspension enclosure represents an intelligent and rather obvious way of getting deep bass out of a relatively small box, but if one is seeking really deep bass in the 20 Hz range, a considerable enclosure volume is still required, anywhere from five to eight cubic feet for reasonable efficiencies.

That's hardly compact, and hardly approaches our ideal everyman's subwoofer.

Bass equalization.

A supremely simple method of extending the bass response of an acoustic suspension system without increasing enclosure size is bass equalization. This can be done passively by means of a network which creates an impedance null below system resonance and causes a solid-state amplifier to dump current into the load and consequently to increase its power output. A better way is to use an active equalizer at line level. In either case equalization buys you bass extension at a very steep price. An acoustic suspension speaker tuned to a Q of 0.7 rolls off at 12 dB per octave. To equalize that 12 dB of droop to flat, assuming the cone has the excursion to keep up with the equalizer, will require minimally sixteen times the amplifier power initially needed to produce a reference level at the unequalized corner frequency. In actual practice, the power requirement is apt to be higher because of a phenomenon called power compression, which afflicts all woofers to varying degrees and arises when the transfer function becomes nonlinear with increasing amplitude. In real-world systems, getting more than a half octave of bass extension through equalization is difficult.

Doesn't look very promising, does it? Nonetheless, bass equalization has been used in a number of consumer products, most notably the Thiel loudspeakers and of course the controversial Bose 901. (The original 901s were equalized acoustic suspension speakers, but later models use equalized ducted enclosures—a special case considered below.) Moreover, the Audio Control "Richter Scale" bass equalizer has been sold for do-it-yourself applications for many years, and the technique is ubiquitous in car audio.

The aperiodic enclosure.

Now let's get a little more sophisticated and consider the aperiodic enclosure, a variation on the sealed box theme utilized at various times by Dynaudio, Fried, and VMPS, among others.

An aperiodic enclosure, also called a line tunnel, is basically a leaky sealed box (see Figure 2) full of damping material. A clear space is left behind the driver, which extends to a sort of occluded port that is very heavily stuffed with damping material held in compression by some kind of mesh. This vent (called a variovent by Dynaudio) functions as a flow resistance, damping the movement of air in the port. The mass of the air in the port is inconsequential in the functioning of this system; rather the compliance of the air in the enclosure dominates, and the system produces only one major resonance, which is more or less completely damped. The air within may be regarded as a heavily damped spring.

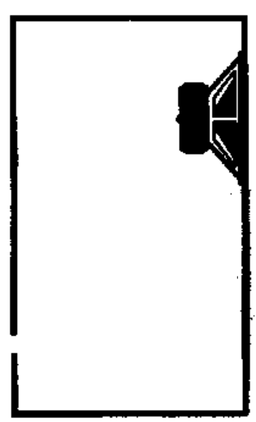


Figure 2: Basic configuration of the aperiodic enclosure.

An aperiodic enclosure is in essence a Helmholtz resonator, but the output of the port is so heavily attentuated and highly damped that box tuning is not nearly as critical as in the case of more conventional ducted-port systems. Practical designers tend to treat the system as a variant of the sealed box and to utilize drivers that have been optimized for acoustic suspension applications.

I should mention that the variovent has also been used in ducted systems, where its function is to lower one of the system resonances—the upper resonance caused by the stiffness of the air in the cabinet.

Because of the resistive leak, the compliance of the air at very low frequencies in an aperiodic enclosure is higher than in a completely sealed enclosure of similar dimensions, and thus the air within tends to behave like a larger volume—about forty percent larger in theory according to Dynaudio applications notes, but nothing nearly as dramatic in actual implementations. *[That "theory" probably came from Dynaudio's adman.—Ed]*

Here one should note that very little has been published about the aperiodic enclosure aside from the Dynaudio notes. I've found no detailed mathematical models, nor even any constructor articles. The following observations are based upon conversations with loudspeaker professionals having extensive design experience with the type, principally with Mike Dzurko of Audio Concepts.

Dzurko's experience indicates that substantial downsizing is not possible with the design. "You're not going to get a lower f_3. What you will get is a very low-Q, very well-damped speaker," he says. "Many misleading claims have been made for this design. We probably have more experience with aperiodic enclosures than any company in the world, and we know what it can do. It will not enable you to make a tiny subwoofer that goes very deep." *[Right on, Mike.—Ed.]*

The compound or Isobarik enclosure.

The next technique, Isobarik loading, is a different story. Isobarik loading can reduce enclosure size requirements by nearly fifty percent for a given 3-dB-down point, provided the inevitable penalty in lost efficiency is acceptable. It is a design that really merits examination if installation space is a primary consideration.

An Isobarik or constant-pressure box may be either sealed or vented, but I'll concentrate on the sealed type.

In the sealed-box Isobarik design (see Figure 3) two woofers are used, one of which actually produces output, and the other of which serves to decouple the working woofer from the air load in the enclosure. The outside woofer is usually mounted on the front panel in conventional fashion, while the inner woofer is mounted at the end of a short tunnel which encloses the back of the outside woofer. The back of the inside woofer works into the larger enclosure.

There are a number of variants of this design, the most common of which has the woofers facing each another

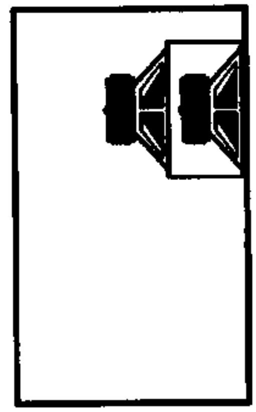


Figure 3: Typical configuration of an Isobarik enclosure.

and wired out of phase, so that the outer woofer fires backwards from the front panel but its cone always moves in the same direction as the cone of the inner woofer. A connecting tunnel is not required in this variant.

Isobarik theory treats the two woofers as a single unit having twice the moving mass of a single woofer of the same type. In terms of extending bass response, there's nothing really magic about the design, but in practice a pair of drivers linked Isobarik fashion will exhibit Thiele-Small parameters which are difficult to achieve in the design of a single driver. A sealed-box Isobarik is in principle an acoustic suspension speaker, one which takes acoustic suspension principles further than is normally the practice, using a very large woofer mass to obtain lower system resonance.

Isobariks are supposed to provide other benefits in addition to reduced enclosure volume, such as lower distortion and better rejection of the back wave, but I cannot comment on the truth of such claims. I can only say that the bass response of a properly executed Isobarik is almost unbelievable on first hearing. Not so long ago I heard a tiny bookshelf Isobarik made by Dynaudio, which had a listed -3 dB point of 25 Hz and used a 6V2" woofer. I am sure that nine out of ten audiophiles on hearing the system would have sworn that a large subwoofer had been concealed in the room.

Nevertheless, the downsizing possible through Isobarik loading entails a couple of penalties. Obviously, a doubling of driver cost is one of them, but not so obviously efficiency is halved. The inner driver in the pair does not produce a useful output, so the power required to move it is essentially wasted. You end up with an enclosure of half the size, but you sacrifice half your efficiency to get it.

I would add that Isobarik loading also creates problems in midrange reproduction at the point where the distance between the drivers equals the half-wavelength reproduced, but in a subwoofer one can pretty much ignore that limitation.

The ducted port.

Historians of loudspeaker design almost invariably cite the publication of the Thiele and Small papers as the signal event that led to high-fidelity commercial realizations

Figure 4: Basic configuration of the ducted-port enclosure.

of the venerable bass reflex speaker, but I tend to think that the introduction of the duct (see Figure 4) in the early sixties was of at least equal moment. Ducts make possible the reduction in enclosure volume to a fraction of what is required for an enclosure using a simple round port. Vented speakers of the fifties were nearly as large as the infinite-baffle designs of the period, whereas modern ducted designs may be smaller than an acoustic suspension system with the same 3-dB-down point.

A duct is not a panacea, and it does impose certain liabilities on a speaker system, but the downsizing it permits is achieved at a remarkably slight cost—which does much to explain the fact that virtually no vented speaker made today uses a simple port.

How does a duct work to reduce enclosure volume for a given f_3 and a given efficiency? Essentially by coupling the driver to a larger air mass than would be the case with a simple port. A vented box is a Helmholtz resonator, and in any Helmholtz resonator only the air immediately within and adjacent to the port imposes a mass reactance upon the system. Most of the air volume within the enclosure forms an acoustic capacitance, that is, a spring upon which the relatively small air mass in and around the port is suspended. When a duct or tunnel is added behind the port, the entire air mass within the duct then exerts mass reactance and, at a certain frequency—the vent tuning frequency—is tightly coupled to the driver by the stiffness of the air within the enclosure. Furthermore, the effective stiffness seen by the driver increases as the vent tuning frequency is approached.

Perhaps at this point we should pause to review the basic operation of vented systems.

All vented systems exploit three resonances to achieve high system efficiency in the bass. The first resonance derives from the total mass of the vent air mass plus the woofer mass, coupled to the stiffness of the woofer suspension. This resonance is very troublesome because it is below the vent tuning frequency, yields no usable output, and serves only to allow large, useless excursions of the woofer. The second resonance is the mass of the vent air load coupled to the woofer by the spring stiffness of the air in the enclosure. The third resonance results from the stiffness of the air in the enclosure plus the stiffness of the woofer suspension resonating with the mass of the woofer and is always higher than the first in a properly tuned vented system.

When a vented system is designed with the correct driver Q and box Q, all of these resonances will combine to form a frequency response that is flat to cutoff, although two electrical impedance rises will be evident to announce the first and third resonances. The parameters of the driver must be precisely tuned to the mass and the compliance of the air in the vent and the enclosure, respectively, to permit the achievement of level frequency response. The duct provides a convenient tool for manipulating the values of mass and compliance.

In a very general sense, the longer the duct, the smaller you can make the box for a given driver, but one quickly runs out of space in which to place the duct, even if it is folded, and one can't very well allow it to jut out of the cabinet because of phasing problems—never mind the cosmetics. In any case, a duct measuring over about eighteen inches will develop severe quarter-wave resonances in the bass region and is likely to cause audible colorations as a result of wind noises in the pipe. If we consider one of the better ducted subwoofers made today, the JBL B460, we will note that internal volume is about twelve cubic feet for a 3-dB-down point of over 20 Hz. Hardly downsized, now is it?

If we're willing to sacrifice efficiency, we may utilize Isobarik loading in a ducted system and reduce box size by almost half. That may result in a subwoofer of quite modest dimensions, rather smaller than would likely be the case with a sealed-box Isobarik. But for 20 Hz response, we're still likely to need something on the order of five cubic feet. And this tactic carries a further penalty—an approximate doubling of required duct length. Again, no free lunch.

The passive radiator.

There exists an interesting and useful variant of the ported box. If a passive radiator is substituted for the duct, a similar reduction in box size is possible, with no additional penalties in efficiency and certain distinct advantages.

A passive radiator—also called a drone cone, though it needn't necessarily be a cone—is a diaphragm without a motor, which is used to fill a nonducted port (see Figure 5). A passive radiator permits a lower tuning from a small-volume box by adding to the apparent mass of the enclosed

Figure 5: Basic configuration of the passive-radiator enclosure. Note that only the active driver has a motor (magnet and voice coil); the passive radiator is only the diaphragm part of a complete driver.

air. The air in the cavity is coupled to the masses of both the driver and the passive radiator, and since the PR can be designed to be very massive, a long air duct can be dispensed with. Typically the diaphragm is weighted down with modeling clay or metal washers, so that it is many times as massive as an ordinary cone, and as such it can simulate the air mass of an enormously long duct. Also, in actual practice, the parameters of a PR can be more consistently controlled than those of a long duct, and the theoretical response profile of the system is therefore more accurately realized.

Passive radiators constitute a fairly popular means of downsizing bookshelf speakers and achieving substantial bass output from a modestly sized enclosure, but in true subwoofer applications they have gained less acceptance. In practice a passive radiator should have at least twice the surface area of the cone driving it, and that inevitably makes for a larger enclosure. PRs also tend to run out of excursion at high drive levels, something a duct can never do. Moreover, PRs are prone to many of the same distortion mechanisms of cones themselves. Unless the PR is provided with a highly linear suspension, it will color the sound of the system, and to linearize the PR with high-quality suspension components is to add substantially to its cost. Also, the correct mathematical modeling of PR behavior is not as familiar to many designers as the standard vented-box model. Accordingly, actual system design is somewhat haphazard. Finally, PRs wear out rather quickly, the movement of the massive diaphragm eventually tearing the surround. I have only seen one consumer component subwoofer using a PR, a rather excellent and very costly speaker called the SL-1, produced by Onkyo several years ago.

In the June and July/August 1981 issues of the AES *Journal*, a variant on the passive radiator called the augmented passive radiator was described by Thomas L. Clarke. The design calls for a kind of passive Isobarik loading with two PRs linked back to back, and with the rear one firing into a subenclosure (see Figure 6). The strategy supposedly provides the outer PR with effectively infinite compliance and results in a reduced front chamber requirement.

Figure 6: One of the several possible configurations of the Clarke augmented passive-radiator system.

But the necessity of having a rear chamber results in an larger box overall than would be the case with a conventional PR system using the same driver and tuned to the same alignment. Clarke suggests ducting the inner PR to inside of a wall or a basement dead space—essentially the same tactic discussed in regard to aperiodic enclosures.

Finally there is the Nestorovic bass system, a sort of variant of the passive radiator, though not exactly that, and in fact not awfully similar to any other subwoofer design.

Nestorovic bass, subject to fairly recent patents filed by Mila Nestorovic, a former Mcintosh and Phase Linear engineer, is available only in speakers marketed under the Nestorovic brand name. The system uses two more or less conventional drivers, one of which is partially decoupled from the amplifier through the crossover network at the deepest frequencies and becomes progressively passive with descending frequency—a sort of demipassive radiator, as it were. Because the passive radiator is not fully passive, its damping and stiffness are higher than is the case with a conventional PR, and thus it permits a range of tunings that would not be possible with a purely mechanicoacoustical system. Nestorovic claims a one-third-octave bass extension over a passive radiator system using the same box size, along with a third-order rolloff characteristic as opposed to the 24 dB per octave rolloff rate which characterizes PR systems. Nestorovic's Type A subwoofer uses two twelve-inch cones and has roughly four and one half cubic feet of internal volume, a claimed 3-dB-down point of 18 Hz, a claimed maximum output of 120 dB at 30 Hz, and a claimed efficiency of 93 dB at one meter with one watt input. I have not tested this system, but I would say that if the claims are true, it represents a significant innovation and constitutes a product which begins to approach the characteristics of the everyman's subwoofer we have envisioned—though at four and half cubic feet it is still a fairly big box. *[I find the claimed performance very hard to believe, unless the trick is to drop the impedance to less than 1 ohm at 18 Hz, in which case it's an unrealistic design.—Ed]*

Bandpass enclosures.

Before I discuss the great family of electronically assisted speaker systems, and the special case of gas bag systems, there is one other low-volume enclosure to investigate: the bandpass or dual-tuned enclosure. This comes in several variants, but in all cases the driver or drivers are mounted internally and modulate a vented cavity or cavities from which the whole output of the system emerges—in other words the bandpass, like the horn, is an indirect radiator. The bandpass is currently arousing a lot of interest in perfectionist circles, though the basic design goes back to the thirties.

Bandpass speakers offer several advantages in performance and some significant drawbacks. A complete discussion of their behavior is beyond the scope of this article, and the only matter I will cover here is the potential the design offers for low bass extension from low cabinet volumes.

In fact, bandpass speakers may be aligned in a great number of ways, only some of which afford low box volumes per a given bass cutoff. The speaker designer has a large number of parameters with which to juggle—

including bandpass width, center resonant frequency, frequency response within the bandpass, efficiency, enclosure volumes, duct length, and duct diameter—and the formulae for doing so are highly complex and have not been fully reported in audio trade journals.

One specific type of bandpass speaker, the compound KEF-type enclosure using one vented chamber and one sealed chamber (see Figure 7), can be tuned to yield astonishingly low box volumes for given corner frequencies; indeed I have seen 30 Hz systems with slightly over a cubic foot of internal volume. Such downsizing is not achieved without significant performance penalties, however. But before I discuss these, I would like to explain how the bandpass configuration enables the designer to effect such remarkable size reductions in the first place.

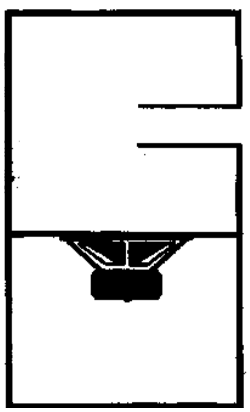


Figure 7: Simplest possible configuration of the bandpass enclosure.

In a KEF-type bandpass enclosure—the simplest sort—the driver or drivers are situated on an interior partition. The tuning of both chambers is quite different from those of conventional sealed or vented boxes in that the system resonances are nowhere near the 3-dB-down points for the bass. Instead, both enclosures will be tuned fairly high—say 40 or 50 Hz—and the two box resonances, which must coincide in a properly aligned system, will be heavily damped so that the resonant peak is flattened over a span of two or more octaves. To put it another way, the two chambers are cascaded acoustical filters.

Each of the chambers, with its individual high resonant frequency, will be small in volume, and it can be shown that the resulting system will have less total volume than a conventional sealed or vented enclosure with the same 3-dB-down point. If Isobarik loading is used along with bandpass configuration, total enclosure volume can be reduced a further 50%. And if a passive radiator is substituted for a duct, the box can be made so small that one has difficulty finding mounting space for the drivers. Intersonics, a pro sound company, built an 18 Hz prototype of this sort with less than three and a half cubic feet of internal volume.

Is one getting a free lunch then?

By no means. First of all, extension is being sacrificed on the top end. That's generally desirable in a subwoofer, but it's still a design compromise. Second, the compound KEF-type enclosure is rather inefficient, as much as 20 dB less efficient than a single-driver bandpass enclosure with ducts in each chamber—though efficiency is inversely related to bandwidth, and by reducing the passband to under two octaves, moderate efficiency may be achieved. Third, the design entails extremely high pressures within the box, with consequent power compression and nonlinear behavior at high drive levels. Fourth, it is very difficult to align for flat response through the passband at any drive level, and nearly impossible to do so at high drive levels. In short, it is a design where everything has been sacrificed to get low bass out of a little box. I say this with reluctance because I spent many, many hours attempting to design such a speaker, hoping that somehow I would evade the usual consequences of downsizing and have myself the everyman's subwoofer we all desire. I failed.

I should mention, however, that the Intersonics prototype, with a volume of under three and a half cubic feet and an 18 Hz corner frequency, boasted efficiency in the 89 dB range. *[I'll believe that when I can measure it for myself.—Ed.]* This was achieved solely by virtue of an extremely efficient, extremely long-throw driver that is powered by a servo motor and is proprietary to the company. Intersonics has considered marketing the design. A commercial version would come very close to our mythical everyman's subwoofer.

Bandpass enclosures with dual or triple ducted cavities can be made to be very efficient and to achieve flat amplitude response over a wide dynamic range, but cabinet volumes tend to be quite large. JBL makes some triple-chamber, triple-ducted Isobarik bandpass speakers that achieve reasonably low bass at high efficiency and at fairly low cabinet volumes. Nevertheless, there's no magic here, and the interior volume of their 30 Hz system approaches ten cubic feet.

We now come to electronically assisted designs.

Sixth-order vented alignments.

I've already touched upon the equalized sealed-box type, the simplest electronically assisted design. Considerably more sophisticated applications of signal processing are possible, some of which will permit very radical downsizing indeed.

The first case to consider is the sixth-order vented alignment described by A. N. Thiele and actually embodied in Electrovoice's long discontinued Interface D. This system design does not represent a brute-force application of line-level equalization; quite the contrary, the amount of equalization required for flat, extended low-frequency response is modest and must never be exceeded, or the speaker may be dangerously overdriven.

Basically here's how such systems work. Most vented alignments in actual use yield flat response through the passband and a sharp 24-dB-per-octave drop in output below the passband. Typical sixth-order alignments in their pre-equalized state more closely approximate Bessel characteristics, with an initial shallow rolloff which steepens into a sharp rolloff with descending frequency. The first octave of droop may be elevated through equalization without overdriving the woofer, and the final shape of the bass response curve will take the familiar Butterworth form. A sharp subsonic filter is generally included as part of the equalization circuitry.

This tactic can yield about half an octave lower extension than could be achieved by a QB3 alignment with the same driver and the same box size. That's not bad, especially insofar as the amount of boost used in sixth-order alignments is at most a few dB.

Nevertheless, the added cost and complexity of an outboard equalizer has deterred speaker manufacturers from adopting this approach, despite its elegance, sophistication, and effectiveness. *[I suspect the reason is rather the additional transient hangover introduced by the sixth-order highpass filter characteristic, which is audible to discriminating listeners unless the f_3 is extremely low.—Ed]*

The ACE-Bass.

This is perhaps the cleverest approach to downsizing ever devised—it really works, and it seems like magic.

The ACE-Bass, a trademark owned by Audio Pro, a Swedish loudspeaker manufacturer, is basically a refinement and extension of an existing technology which had never seen much application in commercial products. In essence it is an electronically assisted vented box, but it is neither an equalized system nor a servo system in the usual sense.

Ever since Edison did his pioneering work on the development of the phonograph, engineers have recognized that mechanical mass and compliance are analogues of inductance and capacitance. An acoustical circuit in series with an electrical circuit, as is always the case in a loudspeaker system, exhibits lumped circuit parameters and combined filter functions. Up to a point, the values of mass, compliance, and acoustical resistance can be augmented or compensated for by inserting electrical values of the opposite reactive property and thus forming a conjugate network for the whole system. This is exactly what is going on in the ACE-Bass.

All ACE-Bass speakers necessarily use dedicated amplifiers which include a reactive network at the output. The amp itself has a negative output impedance, which is essential for the system to work.

Negative impedance is required because the DC resistance of the loudspeaker voice coil virtually swamps the reactance of the amplifier's network. By removing this resistance, the designer can bring the electrical properties of the network into play in shaping system response. For instance, the designer can increase apparent driver mass with a parallel capacitance, stiffen compliance with a parallel inductance, and control damping through a parallel resistance. (The motional impedance of the speaker appears to have been ignored in the published calculations on system operation.)

Let's examine just one particular of system operation: Vented systems as a species are best adapted to very low-Q drivers which themselves permit lower tunings per a given box size; the ACE-Bass system lowers effective driver compliance by driving the woofer with a negative-impedance amplifier and synthesizes a woofer behavior that would be very difficult to achieve by purely mechanical means.

In some sense the ACE-Bass could be said to be an equalized system, inasmuch as it uses a resonant interface circuit to shape response, but because the circuit is conjugate to the acoustical circuit of a given driver, its effects are much more subtle.

The ACE-Bass system is patented and has only been used in commercial applications in Audio Pro brand speakers. The results—sad to say—have not been extremely encouraging. The systems all use small drivers of limited output capabilities and unremarkable performance, loaded into cabinets which are far from acoustically inert. The efficiency of the system is difficult to judge because it demands a dedicated amplifier, but I'm convinced that it is low; any time you ask an amplifier to drive a highly reactive load, it's going to deliver less usable power into the load and more into the heat sinks—and the design necessarily incorporates a reactive network between amp and speaker terminals. The AES paper that introduced the ACE-Bass system back in 1981 indicated a system efficiency of only about 85 dB, one watt, one meter. So essentially we're back to the same trade-off—efficiency against box size. *[See Issue No. 9—then called Vol. 2, No. 3, published in 1980—where my review of the Audio Pro B2-50 agrees with the author's disappointment in the commercial implementation.]*

Active servo technology.

Introduced by Yamaha a couple of years ago, this is a system that apparently has much in common with the ACE-Bass. The system is similar mainly in that a negative-impedance amplifier is used. A current sensor measures voltage drop across the voice coil resistance, and this is then compared with the amplifier output. Output voltage is then boosted to compensate, thus producing the negative impedance.

No conjugate network is used, and the sole parameter manipulated is loudspeaker damping, though by this means Yamaha does manage to achieve impressively low corner frequencies in small cabinets. I have no hard evidence on system efficiency.

Motional feedback.

Motional feedback is a technology that is forever being rediscovered. First proposed back in the nineteen twenties, it appeared in consumer products in the nineteen fifties, disappeared for almost a decade, and reappeared in the Infinity Servostatic in 1968. Infinity Systems has used the technology in selected products ever since, and more recently such companies as Entec, Velodyne, and Canton have offered speakers with servo-controlled bass.

In essence motional feedback is very simple. Cone motion is sensed in any of a number of ways, and the transducer converts the cone motion into a varying electrical voltage which is then fed back out of phase into the driving amplifier. The nonlinearities of cone motion will thus be canceled, while at the same time amplifier gain will be reduced. The process and its results are indistinguishable from ordinary global negative feedback as applied to power amplifiers. The only difference is that the speaker itself has been included in the feedback loop.

Motional feedback, in most instances, will substantially decrease loudspeaker distortion, optimize transient response, and will also extend bass response. Here I will only concern myself with the last benefit. Motional feedback in all its manifestations is simply too large a subject to consider otherwise.

If one's main concern is to extend bass per a given cone and box size, the most direct means of sensing cone motion and converting it to a useful feedback signal is to utilize an accelerometer. As you will recall, cones must maintain constant acceleration with descending frequency to maintain flat response. An accelerometer will detect the point where acceleration drops off, and the antiphase feedback signal will cause the driving amp to increase output to make up for the natural loss.

In terms of its effect on amplitude response, a motional feedback system is little different from an equalizer. It has the same inherent inefficiency and thus demands an increase in amplifier power. As with equalization, motional feedback entails the danger of overdriving the cone, and moreover such systems are prone to instability and destructive oscillation. Designing a good, safe motional feedback system is not a trivial engineering project, hence the rarity of such technology in real-world products.

Why use motional feedback over simple equalization? We mentioned reduced distortion, though in fact bass equalization can significantly reduce second harmonic distortion as well. The real reason for preferring feedback over simple equalization is more subtle. Motional feedback is a particularly elegant, if rather complex, way to improve damping and transient response in addition to reducing distortion and flattening frequency response. That is its point of superiority. On the other hand, feedback offers no improvement in efficiency whatsoever. It shares with equalization the same efficiency penalties in the trade-off against extended bass response. You can't get away from the "eternal triangle."

Gas bag enclosures: Rube Goldberg ascendant

To those who have not troubled to explore the less traveled byways of esoteric audio, the idea of stuffing blimps into a speaker enclosure to reduce volume must seem entirely preposterous. Nevertheless, this tactic, while of somewhat dubious practicality, does yield results and does bear discussion.

Basically the gas bag school is divided into two camps. The first camp, whose sole inhabitant is Michael Day ton-Wright, a Canadian inventor, simply exploits the differing densities of various gases. The speed of sound depends upon the density of the medium; higher density means higher speed. Certain flourocarbons are characterized by lower density than air, and when such gases are placed in an enclosure, the enclosure itself will resonate at a lower frequency. The gas is easier to compress than air, and the woofer "thinks" it sees a larger volume. A system using gas-tight bags of flourocarbons can certainly be made to work and has seen application in real commercial products. There is always the problem of leaks, however.

The second approach, first advocated by Gene Czerwinski of Cerwin-Vega and later by an ex-IBM mechanical engineer named Ralph Marrs, exploits the fact that certain flourocarbons, such as Freon vapor, liquefy under pressure with a sudden significant loss of volume. The process occurs in under a millisecond, which means that a woofer will see a large vacuum behind it at the peak portion of a wave cycle.

I have interviewed Mr. Marrs in his laboratory, seen his demonstrations, purchased and used his gas bag modules myself, and I can attest that they perform as advertised. One gas bag, taking up less than a liter of cabinet volume, effectively reduces cabinet volume requirements by four liters in a sealed enclosure. In some instances enclosure volume can be reduced more than fifty percent, and at no loss of efficiency!

Is there a drawback?

Isn't there always?

Air inevitably leaks into the modules, reducing their effectiveness over time. Projected half-life is about five years. And because the modules affect only a single box-tuning parameter, namely the compliance of the air volume, and because the conversion process itself is classically nonlinear, the modules are of dubious utility in reducing the volume requirement of a highly tuned vented enclosure. The gas bags also very expensive.

Still, it's a remarkable technology and remains the only way to get around the "eternal triangle"—by cheating a little and changing the atmosphere inside the box.

So, let's see now...

Suppose we are still determined to have a 20 Hz system taking up no more than one cubic foot of enclosure space. How would we build it?

We'd start with a KEF-type compound configuration, with two woofers face-to-face on an interior partition. The back chamber would be sealed and the front chamber vented with a passive radiator. The rear chamber would be stuffed with Marrs supercompliant modules, and the drivers themselves would be controlled by motional feedback. The system, in fact, could probably be made to dimensions of less than a cubic foot. It would be enormously inefficient,

Summary: Box Volume Reduction Strategies for Subwoofers

ACOUSTICAL METHODS	EXAMPLES
1. Acoustic suspension or air spring enclosure. Uses cone with very low tuning (due to high driven mass and high compliance), which in turn lowers tuning of small sealed box.	Ubiquitous enclosure type used by such manufacturers as Alison Acoustics, Acoustic Research, Advent, Cambridge Soundworks, and many, many others.
2. Isothermal stuffing. Absorbs heat produced by the compression of air in the enclosure, thereby lowering the stiffness seen by the woofer and increasing the effective volume of the enclosure.	Ubiquitous construction technique used in virtually all high-quality acoustic suspension speakers.
3. Internal gas bags. May use one of two different methods of increasing the compressibility of the sound medium and thus the compliance of the enclosed volume.	One method has been used in Dayton-Wright subwoofers, the other in prototypes made by Marrs Development.
4. Aperiodic enclosure. Low-Q variant of the sealed box. Lowers tuning by absorbing energy at the resonance peak; saves magnet size and expense.	Used in VMPS subwoofers.
5. Isobarik enclosure. Lowers tuning by effectively doubling cone mass.	Used in Dynaudio and 3A Audio Design subwoofers.
6. Ducted port. Lowers tuning by increasing air mass coupled to driver below vent frequency.	Ubiquitous enclosure type, used in JBL subwoofers (among many others).
7. Passive radiator (PR). Operates on essentially the same principle as ducted port.	Used in Onkyo SL-1 and Intersonics professional subwoofers.
8. Augmented passive radiator. Lowers PR tuning by making PR compliance effectively infinite.	Never used in a commercial product.
9. Nestorovic bass. Employs electromechanical manipulation of PR operating parameters and increases drive current at very low frequencies.	Used in Nestorovic brand loudspeakers exclusively.
10. Bandpass enclosure. Indirect radiator with resonant chambers tuned to achieve a wide separation between center resonant frequency and low-end corner frequency.	Used in Bose, 3A Audio Design, Kenwood, and JBL subwoofers.

ELECTRICAL METHODS	EXAMPLES
1. Bass equalization. Brute force method for extending the low-frequency corner by boosting amplifier output.	Thiel and KEF loudspeakers.
2. Sixth-order vented alignment. Uses a moderate amount of boost to "fill in" the highpass filter characteristic of a slightly overdamped ducted port system.	Electrovoice Interface D, B&W Matrix 801 Series 2.
3. **The** ACE-Bass. Electrical synthesis of mechanical parameters to lower system tuning.	Used in Audio Pro subwoofers exclusively.
4. Servo bass technology. Electrical synthesis of one mechanical parameter, namely damping.	Yamaha subwoofers.
5. Motional feedback. Inclusion of the woofer cone within a feedback loop that applies error correction to the drive signal.	Infinity, Entec, and Velodyne subwoofers.

however—perhaps in the mid-70s dB range—and peak output would probably be only moderate, perhaps a bit over a 100 dB. Interior box pressures would be extraordinarily high, and system lifespan would be short—all in the interest of downsizing.

I for one would very much enjoy building such a system.

But I wouldn't care to own it. •

Four More Multibit D/A Processors

By Peter Aczel
Editor and Publisher

This could be the last flowering of the multibit DAC technology before a complete delta-sigma takeover.

A so far not very thorough look at the latest generation of delta-sigma (one-bit, bitstream, whatever you want to call them) DACs gives the initial impression that they outperform even the best multibit designs on the test bench. The following reviews of "flagship" D/A processors from three different makers who have so far stayed with the multibit technology should be read with that in mind.

Please note that the reviews don't go into sonic comparisons of these units. David Rich put three of them—the Aragon, the EAD, and the smaller PS Audio—through a round-robin of ABX listening tests and reported no audible differences whatsoever. That didn't surprise me, since the measurable differences were by and large far below the known threshold of audibility. Individually, all four units sounded flawless to me.

Aragon MKII D2A

Mondial Designs Limited, 2 Elm Street, Ardsley, NY 10502. Aragon MKII D2A outboard D/A converter, $1595.00. Tested sample on loan from owner.

The D2A unit reviewed in Issue No. 15 was sent back to Mondial Designs for updating to the MKII version and then retested. The only difference between this modified unit and a brand-new MKII D2A was that the original power supply had not been replaced; the new, larger IPS power supply, formerly an option, now comes with all newly purchased MKII's, unless you specifically request the old one to keep the price down to $1295. As David Rich explained in the last issue, the older supply is perfectly adequate.

The changes introduced in the MKII D2A aren't trivial. The fully discrete class A analog stage now has a low-impedance FET-buffered (source follower) output, so that the impedance of the load has become uncritical, and the differential pairs are now biased by current sources. Thus the objections raised in Issue No. 15 are gone. Furthermore, the digital section incorporates a new circuit that further reduces jitter in the recovered clock. These updates are available to all owners of the original D2A.

Unfortunately, the net result of all these intelligent changes is that only dumb little errors remain—errors that are avoidable even in moderately priced equipment, let alone a high-end D/A processor. The retesting of the MKII on the Audio Precision System One—which has a much greater full-scale display range and better amplitude accuracy than the sweep spectrum analyzer I used to test the original D2A—revealed the errors only too clearly. Once again the MSB trim pots were incorrectly adjusted, resulting in far from negligible gain linearity errors at the -80 dB and -90 dB levels (approximately -2 dB and -3.5 dB, respectively, in both channels). Even the -70 dB reading was off by almost 1 dB. That's ridiculous. Obviously Mondial Designs doesn't have the test equipment to adjust the trim pots accurately. With the Audio Precision it took me about two minutes per channel to reduce all deviations to less than 0.5 dB at all levels down to -100 dB. With my old spectrum analyzer it would have taken a little longer. Come on, Mondial Designs. You can do what I can do. Or use the K-grade DACs without any trim pots to avoid the problem in the first place—whichever solution costs less, one or the other. One thing is certain: the dealers don't know how to adjust the MSB trim pots, and even if they did they wouldn't.

There was also another little boo-boo, which may have been the result of a deliberate trade-off, so perhaps I should give it the benefit of the doubt. The THD + N measurement at any frequency below 5 kHz with a full-scale input *would have been* between -92 and -94 dB, which isn't half bad, but a spurious 60 Hz component made it worse by about 6 dB. That indicates a ground loop, since the completely separate power supply is obviously not leaking 60 Hz into the main chassis. Now, the grounding scheme of the Aragon has the elimination of RFI as its top priority, and indeed the unit is the champion of them all in that respect—no radiated RFI whatsoever, meaning no TV or FM reception problems in its proximity. The small ground loop may have been the design trade-off against that major advantage. It's well below audibility in any event.

All other measurements were close to perfection, surpassing both PS Audio units and approximately equaling the EAD (see reviews below). The general endorsement of this basic design in Issue No. 15 still applies, especially after the various circuit improvements. The Aragon MKII D2A is a very competently designed D/A processor made with very high-quality parts; if Mondial corrected that one easily correctable little flaw (or is it two flaws?), an unqualified recommendation would be in order, especially at the lower price with the original power supply.

EAD DSP-7000

Enlightened Audio Designs Corp., 607 West Broadway, Fairfield, lA 52556. DSP-7000 outboard DIA converter, $1299.00. Tested sample on loan from manufacturer.

This is the unit previewed in the last issue, at the end of my review of the EAD "AccuLinear" mods. Now that I've had my hands on the production DSP-7000, listened to it, and measured it on the Audio Precision System One, I can report that it ranks very high in the current hierarchy of multibit D/A processors, probably as high as any. It's a well-thought-out, scientifically engineered, full-featured, beautifully built piece of equipment; however, the claimed superiority of the AccuLinear I-to-V converter to all other design approaches was not evident in my tests. More about that in a moment.

I was impressed by the construction of DSP-7000. It's rather compact yet it weighs a ton (21 pounds to be exact) because of the heavy-gauge chassis, massive toroidal power transformer, and almost half-inch thick front panel. The latter is 24-karat gold-plated with a mirror finish to make the DSP-7000 stand out from the competition; I find this touch to be in dubious taste and aesthetically incompatible with the high-tech context—how would you like a gold-plated computer? The mirror finish is also extremely sensitive to nicks and fingerprints; a black anodized front panel will be a future option, I'm told.

The silicon in the DSP-7000 is all pretty advanced stuff. The S/PDIF decoder is the quite new CS8412-CP EP from Crystal, originally recommended to EAD by our own David Rich. The digital filter is the standard but still good NPC SM5813APT, chosen because it's the only game in town when you want a 20-bit output word length, the DAC being the new 20-bit Analog Devices AD1862N-J. (The J grade is just as good as the K grade, EAD tells me—yeah, right). The I-to-V converter is a chip inscribed with the AccuLinear name and nothing else; I understand it's made for EAD by Analog Devices and it looks exactly like the additional AD841JN used at the output. (It also has exactly the same pinouts.) The AccuLinear stage probably incorporates the first stage of filtering; the AD841 stage the next two. EAD has so far resisted providing any further information on AccuLinear. An educated guess is that it's nothing more than a very good voltage-feedback I-to-V converter stage using some kind of fast op-amp. Construction quality under the cover of the DSP-7000 is very fine throughout.

The user features of the EAD are equally impressive. There are eight "idiot lights" on the front panel; I've never seen that many on an outboard D/A processor. Three digital inputs are provided, each with its own button and own light: coax I, coax II, and optical. There's also a coax digital output (for recording on DAT, etc.). Another button/light is for polarity inversion, still another for standby/full power. In the standby mode the light glows dimly as long as the unit is plugged in; the digital inputs and analog outputs are muted, while the active components are kept warm. When the button is depressed the light glows brightly and all systems are go. As soon as a digital input source is selected the lock light comes on; any uncorrected digital error will activate the red error light; pre-emphasized CDs will turn on the de-emphasis light. It's more fun, as David Letterman likes to say, than humans should be allowed to have. For even more fun, take off the cover and play with the slide switch that sets the digital filter to either 4x or 8x oversampling. Each setting has its theoretical advantages; I found no measurable or audible difference. As a $399.00 option, EAD also offers balanced outputs on the DSP-7000 (in case your preamplifier has balanced inputs). A further option, for an additional $249.00, is the AT&T glass-fiber optical interface. If you've read David Rich's comments on that subject in the last issue, you probably won't order it.

The measured performance of the DSP-7000 was outstanding but no better, everything considered, than that of a properly adjusted Aragon MKII D2A, which claims no proprietary technology. One of the most important measurements (which I never see in either the high-end or the popular magazines except *Audio)* is THD + N versus frequency at full scale (0 dB). The DSP-7000 hovered between -89 and -91 dB at most frequencies, but that included the contribution of a spurious 60 Hz component. Without the latter this measurement would have been approximately 2 dB better at the corresponding frequencies. The 60 Hz boo-boo was confirmed by the clearly visible 60 Hz modulation of a 1 kHz tone at -90 dB when observed in the time domain. The 10 kHz distortion was very close to that of the Aragon, indicating very similar jitter performance (i.e., good). Low-level gain linearity was just about perfect down to -90 dB and off by only 0.5 dB at -100 dB; that's with the J-grade DAC and no MSB trim pot, indicating that 20-bit architecture has something to be said for it. The noise floor with a 60 Hz notch filter was between -95 and -96 dB; without such filtering it was worse by 3.5 to 4 dB. All that paints a very decent picture, but there was no evidence of any kind that the proprietary AccuLinear circuit was doing something special.

Now for the clubfoot of the DSP-7000, namely radiated RFI. That's not the same thing as RF coming out of the analog output; from that point of view the unit is virtually perfect, about the best I've seen. But if you're using the DSP-7000 and somebody in the next room is trying to watch TV with an indoor dipole antenna (rabbit ears), be prepared for some angry protests. The TV signal will be snowed under, especially on the lower channels.

All in all, this is a very good D/A processor and quite acceptably priced considering all the goodies in it. Anyone who buys it with full awareness of what it is and what it isn't will be a "happy camper" in my opinion. But if you don't go in for big gold cuff links, gold pinkie rings, and gold tiepins with your initials on it, you'll probably want to order the black front panel—if indeed that option is made available by EAD.

PS Audio Digital Link II

PS Audio, Inc., 302 South 13th Street, Grover City, CA 93433. Digital Link II outboard DIA converter, $799.00. Tested sample on loan from manufacturer.

Already mentioned in our last issue in David Rich's quickie update on CD playback equipment, this face-lifted successor to the original Digital Link has meanwhile been tested on the Audio Precision System One and auditioned in my listening room. The face-lift improved both the image and user convenience of the equipment, as well as its perceived value at an unchanged price; the more rigorous measurement protocol, on the other hand, tended to downgrade the unit slightly in my esteem.

The Digital Link II now comes in a neat black chassis with handles, LED indicator lights, and tiny touch-sensitive switch buttons for on/off and coax/optical source selection. Unfortunately, you must have slightly sweaty hands for the switches to work; if you have dry hands as I do, you need to moisten your finger slightly before operating the buttons, otherwise the relays might not respond. (It is possible that business conditions at PS Audio at the time the switches were finalized were such that everyone was sweating, so the problem went unnoticed.) The power supply is still housed in a separate unit, with an "umblical cord" connecting it to the main chassis. The circuitry remains essentially unchanged, except that the 18-bit DAC used is now an Analog Devices chip (AD1860N-K, the highest grade) instead of a Burr-Brown. Still no MSB trim pots—good! One problem that hasn't been fixed is radiated RFI (see the EAD review above—the PS Audio is no better).

The more precise measurements confirmed the previous finding of almost perfect gain linearity down to -90 dB (and then some), but the frequency response was found to roll off by 0.2 dB at 10 kHz and 0.7 dB at 20 kHz (both channels), a bit more than is desirable—or was PS Audio striving for a barely discernible softening of the extreme highs? It's a minor point, in any case.

The bad news is that the residual noise and distortion turned out to be worse than I had been able to see on a sweep spectrum analyzer. The latter shows the noise floor as a continuously swept, frequency-by-frequency display; whereas the Audio Precision gives—among many other things—an integrated reading for THD + N. That reading was -84.5 dB for a full-scale input at 1 kHz and -88.0 dB for inputs below -3 dB. That's far from 16-bit performance. I knew all along that the theoretical advantages of a passive (resistor) I-to-V converter and of a fast-settling, high-slew-rate op amp like the AD847JN—each a feature of both Digital Links—came with a noise penalty, but now I'm beginning to wonder whether this much of a trade-off is acceptable. Furthermore, the THD + N versus frequency curve at full scale showed a sharp rise above 1 kHz; at 10 kHz it reached -77.5 dB. That indicates a jitter problem.

The twin-tone CCIF intermodulation distortion test (13 kHz + 14 kHz) also gave less clean results than in the case of the Aragon MKII D2A, for example. The 12 kHz and 15 kHz sidebands were at -81 dB, and between 1 kHz and 3 kHz there was also a bit of "tall grass." This is most probably due to the same nonlinear characteristic as the difference in THD + N between full-scale and reduced inputs.

In listening, the noise floor was audible in the quietest passages if one carefully turned up the volume to an abnormal level in those passages and focused on the hiss; I can't say it was even marginally significant. That would be the only way, according to my lights, to distinguish the unit sonically from other high-quality digital playback equipment. Overall, the original endorsement of the PS Audio Digital Link design by this publication must now be toned down to "it's pretty good considering the price but it could be better with just a few little changes." Even so, the Digital Link II remains the lowest-priced outboard D/A converter available.

PS Audio Superlink

PS Audio, Inc., 302 South 13th Street, Grover City, CA 93433. Superlink outboard D/A converter, $1195.00. Tested sample on loan from manufacturer.

This product doesn't make a whole lot of sense to me at its price. I have the impression that someone at PS Audio asked, "How can we make more money with the Digital Link design?" and someone else (was it Bod Odell, since Paul McGowan is gone?) came up with the Superlink.

What's a Superlink? It's a Digital Link II with (1) a deeper chassis, (2) three power supplies inside the chassis instead of one power supply in a separate housing, (3) an additional touch-sensitive button for Invert, (4) additional LED indicator lights for Data Stream, Pre-emphasis, and Copy Guard, and (5) the Burr-Brown OPA602AP op amp at the output instead of the Analog Devices AD847JN. That's all. The Burr-Brown has a longer settling time and a lower slew rate than the AD but somewhat lower noise; it also has higher open-loop gain and lower closed-loop distortion so that the slight nonlinearity of the Digital Link II at 0 dB can be eliminated. But a 50% price increase for these small differences? With no significant improvement in performance? I can't see it.

The Audio Precision measurements were virtually identical to those obtained on the Digital Link II. The main difference was a little less third-harmonic distortion with a full-scale input at 1 kHz, resulting in a 4 dB improvement in the integrated THD + N reading in one channel, which was almost completely negated in the other channel by a 60 Hz feedthrough (three power supplies notwithstanding). The "jittery" rise in distortion at higher frequencies was also comparable; the 10 kHz reading was -80 dB. So much for your extra $396.

Unless you simply can't resist the extra bells and

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A Quick Look at Two Cassette Decks with Dolby S Noise Reduction

By David A. Rich, Ph.D.
Contributing Technical Editor

While the high-end snobs weren't looking, the humble cassette deck grew into a pretty sophisticated piece of audio equipment.

What are we doing reviewing cassette decks in a high-end audio publication? That is what the audiophiles are saying at this point. The music lovers among you already know the answer: a tape recorder is the only way to capture live-on-tape broadcasts on FM. Often these performances are better than what is available on commercial recordings, and often works unavailable commercially are played on these broadcasts. A tape recorder is also the only way to copy an out-of-print recording. (Do not tell me about copyright restrictions. Under my rules, if I cannot buy a new copy of the Bernard Herrmann Symphony—a masterwork of the 20th century that would be much better known if Herrmann were not also a film composer—from Unicorn Records, I have the right to copy it.)

So what medium do we use? Open-reel tape has the advantage of long recording time and is capable of excellent recordings at 7.5 ips using dbx noise reduction. The downside is that open-reel tapes are large, cumbersome, and expensive. DAT makes the best recordings but it never became a consumer medium—try and purchase a DAT recorder at a mass-merchant retailer or a DAT tape at your local record store—and professionals are still using the more robust U Matic format for mastering. That leaves the semiprofessional users and I do not know if enough of these users exist to prevent the DAT format from going the way of the El Cassette. The introduction of DCC is now delayed until late 1992. The latest word from industry sources is that the delay is caused by the difficulties of mass-producing the thin-film recording heads used in the DCC system. So at the present time the medium of choice is the lowly cassette.

To bring new life to the format Dolby Laboratories has developed a new noise reduction system more complex than the current Dolby B and C. This system, called Dolby S, was described at length in the June 1990 issue of *Audio,* so details will not be repeated here. The system is claimed to offer lower subjective noise levels and allow sources with wider dynamic range to be recorded. The highlights of the system are the introduction of low-frequency noise reduction without the pumping problems of dbx and much better portability between machines than Dolby C. It is also claimed that a Dolby S tape can be played back on a Dolby B machine without significant sonic degradation. Dolby gave the job of designing the integrated circuits for Dolby S to—of all people—Sony. You remember Sony, the company that developed DAT during this time period. And—surprise!—Sony took a much longer time to develop the chips than Dolby expected. Now I am not claiming a conspiracy here. I am just making an observation. Custom analog LSI circuits are often late because of unforeseen problems that only show up after silicon is produced. Each revision of silicon takes three months or more. With such a long cycle, development times can slip very easily. But I cannot understand why Dolby did not let Signetics or National Semiconductors, who had done an excellent job in developing the Dolby B and C chip sets, develop the Dolby S chip set.

To check out the state of the art in cassette decks, I decided to examine two Dolby S cassette decks form Pioneer and Harman/Kardon.

Pioneer Elite CT-93

Pioneer Electronics (USA) Inc., 2265 East 220th Street, P.O. Box 1720, Long Beach, CA 90801-1720. Elite CT-93 stereo cassette deck, $1200.00. Tested sample on loan from manufacturer.

This is the top-of-the-line cassette deck in the Pioneer Elite line. It sells for $1200.00 and looks it, with walnut side panels and a polished black aluminum front panel. The unit has a list of features a mile long. Among the highlights are a pressure-pad release which removes the el cheapo pressure pad in the cassette from the the tape head, a 210 kHz bias oscillator, and long-lasting "amorphous" heads (tape heads are expensive if you have to replace them). The tape mechanism is as large and robust as anything I have seen, with a giant flywheel. Pioneer claims for this mechanism a weighted wow and flutter of 0.022% with a 3 kHz test frequency. That is the same figure as my open-reel tape recorder's specification. How can this be achieved with the tape running at ¼ the speed and having ½ the width? It turns out that it has to do with the weighting and test frequency used. A 15 kHz signal was recorded with much less amplitude variation on my old open reel deck (Teac X-2000R) than the CT-93.

The big star feature of the CT-93 is the automatic bias, equalization, and record level tuning system that Pioneer calls Super Auto BLE (I just love the names that the marketing types invent). The bias is adjusted by a motorized potentiometer. It is a nerd's delight, since you can actually watch the bias pot turn during calibration. But the big deal here is that 3 calibration tones are used (400 Hz, 3 kHz, and 15 kHz), that the level adjustment has a large 32-step range, and that the equalization is also adjusted. A 16-step adjustment is made at 3 kHz, and a 3-step adjustment is made at 15 kHz. If you listen to a tape after calibration, you can hear the tone generator rapidly switch between the 3 frequencies as the system interactively tunes in the four adjustable parameters.

I just love this thing and I used it every time I made a recording. Press the BLE start button and the bias knob entertains you for about 30 seconds; then the tape is reset to the original starting point and you are ready to record. No more worrying about tape-to-tape variations. No more concern about when the tape your cassette deck was calibrated for will be discontinued. No more fiddling with bias and level adjustment pots. My frequency response measurements showed the system works very well. Every high-quality tape I used with this machine was adjusted to have flat high-frequency response with excellent high-frequency extension. On some tapes a saddle or a peak of about 1 dB in amplitude would occur between the test frequencies of 3 kHz and 15 kHz, but this is a very minor problem.

The ergonomics of this machine are very good. Tape motion controls worked smoothly. I could never get the unit to dump the tape. Mechanical noise is very low. The CT-93 has a large 16-segment digital level display. Press the meter range button and it indicates levels from -3 to +12 dB in 1 dB steps to aid in setting levels accurately. The meter also has a peak hold mode. Nice as this display is, I would rather have a pair of nice big analog meters so you could resolve 0.25 dB steps. The tape counter has three modes: normal tape count, time count, and a remaining-time counter. The counter always starts out, on power up, in the normal count mode. The recording time of the tape must be manually entered before you can use the remaining-time counter. For the next model it would be nice if the unit could remember the user-preferred settings of the tape counter so you would not have to play with the counter controls each time the unit is powered up. For a touch of whimsy, the open/close button actually opens and closes the cassette door, using a motor like a CD player. For one-step operation the door will also close when a tape-motion button is selected. The mechanism for the door motion control is quite complex and it failed on the first sample of the CT-93 supplied by Pioneer. The MPX filter is at the rear of the unit, where it is easy to forget which position it is set to. The monitor switch automatically goes from source to tape when the record or play mode is selected. I would rather have total manual control of this switch.

The instruction manual could be much better. It covers not only the CT-93 but two other quite different cassette decks made by Pioneer. Operation of some controls is not well explained. For example, you can defeat the Dolby HX Pro adaptive bias adjustment but the manual never explains why you would want to do this. Do not ask me; I have no idea why you would want to do it.

Enough already—what does it sound like? I brought out my killer CD for testing noise reduction systems. This is the second movement of the Debussy string quartet (and you thought it was going to be *The Rite of Spring).* For those of you have forgotten about this movement, I quote from Melvin Berger's *Guide to Chamber Music:* "The second movement offers a profusion of sparkling tonal effects, led by the viola playing an obstinately repeated, quickened version of the motif. Above, beneath, and all around this ostinato figure, the other instruments furnish brilliant pizzicato flourishes and scintillating cross-rhythms." Pizzicato against bowed strings is a good test of a noise reduction system because the decoder must adapt very quickly to this complex set of waveforms. The pizzicato passages were slightly dulled. A very small loss of overtones was observed in the bowed passages but the string tone did not become strident. The "sparkling tonal effects" were reproduced with surprising fidelity. I used the Malcom Arnold Symphony No. 7 (Conifer CDCF 177, a great work of the late 20th century) to see how more complex material was recorded. As expected, it was harder to tell the difference with a mass of orchestral sound. Complex sections were produced with only minimal loss of detail. Bass was reproduced accurately. I used the Julius Katchen recording of the Brahms 3rd Sonata (London 430 053-2; this is a wonderful piece when performed by a master pianist such as Katchen) to test pitch stability. Pitch stability was shown to be excellent. All tests above were run on Maxell XLII-S tape, which is not a recommended tape for this machine. I did this to see how well the BLE system would work. Note that Maxell XLII-S is a relatively inexpensive Type 2 tape, not an expensive metal tape.

If you end up with the impression that I had more fun testing the CT-93 than the 8 preamps I have tested so far for the forthcoming survey in the next issue, you'll be right. This is the best cassette deck I have ever used. It is sonically comparable to the Teac X-2000R open-reel deck, which now costs $2350.00. You do not know how much it pains me to make that last statement. Highly recommended.

Harman/Kardon TD4800

Harman/Kardon Incorporated, a Harman International Company, 8380 Balboa Boulevard, Northridge, CA 91325. TD4800 cassette deck, $1199.00. Tested sample on loan from manufacturer.

This unit is similar to the CT-93 and sells at the same price. Unlike the CT-93, it includes a remote control in that price. I have had this unit in my house for a year. It should have been reviewed in the last issue, but I procrastinated writing the review. I also wanted to compare it with the Pioneer CT-93. I did not get a working sample of the Pioneer unit until recently. Throughout the year I used the TD4800 as my principal cassette deck. I would estimate that I gave the unit at least five hours of use per week. So you might consider this a long-term test like the ones the car magazines perform. The TD4800 performed flawlessly throughout the year. This is also typically the experience of others who have used Harman/Kardon cassette decks. In contrast, the Pioneer CT-93 had a door mechanism failure after a one-week stay with me.

Now for the bad news. The ergonomics of this unit are very poor. For starters there is no record button. The record button is combined with the pause button. If the unit is in the play mode, pressing the record/pause button causes it to pause. If it is stopped, pressing record/pause button will put the deck in record-ready mode. You then press the play button to record. If you press record/pause and play *together*, you get play, not record. I lost several recordings because of this dim-bulb idea. Now let us say you want to check if a tape is blank before you record on it. You play the tape briefly before you record. With the the TD4800 you must be very careful how you do this. If you go too quickly from play to stop to record/pause to play, you will be in the play mode and not the record mode, and your recording will be lost.

The eject button is not interlocked with the stop button. Press it during record or play mode and you can damage the tape mechanism. Press eject during fast forward or rewind and you can spill the tape. You cannot view the cassette well on the TD4800 because it is not lighted and the plastic on the door is smoked plastic. So you use the remaining-time counter to find out if the tape is near the end, right? Wrong. The TD4800 does not have this feature. You have to use the time counter, but you must remember to reset the counter at the beginning of the tape if the information is to be useful. One nice feature of the TD4800 is that the time counter operates in the fast-wind modes. The Pioneer does not account for a time change during fast wind. Like the Pioneer, this unit has an automatic tape monitor. But, unlike the Pioneer, the unit will go into source mode when the tape runs out! It is possible to think you are still recording and listening to the tape when the tape has stopped and you are actually listening to the source!!! This would never happen with a manual tape monitor. I lost part of a recording because of this. A further problem with the monitor switch is a delay after the monitor button is pressed until the signal sources are actually switched. To make A/B comparisons the tape monitor switch on the preamp must be used. Another difference from the Pioneer is that the plastic front panel and controls do not convey a sense of luxury.

My $200 Aiwa cassette deck has a very useful, simple feature called cue/review. Cue/review makes the tape audible during fast-wind operation so you can quickly find the next selection on the tape. This is accomplished by simply moving the play head close to the tape when cue/review is activated. On the TD4800 we get a high-tech solution to the problem of locating things on a tape, called skip forward/reverse. The idea is that the cassette deck identifies pauses between recorded sections and skips to each pause. It did not work at all well for classical music. It would keep stopping in the middle of movements for unknown reasons. The Pioneer CT-93 also had skip forward/reverse (they call it music search) but it was even worse. Sometimes it would get stuck at one section of the tape and not advance at all. Skip forward/reverse—an example of technology marching backwards.

The level meter of the TD4800 has only 12 segments. It does not have an expand mode. You must adjust the bias and record levels manually. Equalization cannot be set at all. Because the level meter has only 1 dB steps at the calibration level, the bias and record level cannot be set with great precision. Howard A. Roberson did a complete set of measurements on this unit for a review in the February 1991 issue of *Audio.* His measurements show mostly excellent performance. My frequency response measurements confirmed his findings, including some abnormalities around 40 Hz. These may be due to a phenomenon called head bumps, which can occur when the tape head is not shaped optimally. I found the bass of the TD4800 to be slightly less detailed and defined than that of the Pioneer. The measured frequency response deviation in the bass is the most likely cause of this problem. Top-end frequency response measurements showed only a small degradation after my year of use of this unit.

Compared to the Pioneer CT-93 the tape mechanism of the TD4800 does not appear to be as robust. The flywheel is less massive. The spindle on the main motor which drives the flywheel through a rubber belt is made of plastic, not metal. Mechanical transport noise was higher than average. On the positive side, the mechanism which detects the record safety and tape type sensor holes is much less complex and more robust-looking on the Harman/Kardon. In addition, the complex motorized door mechanism that failed on the Pioneer is not present on the Harman/Kardon. The Harman/Kardon uses discrete low-feedback op-amps for signal processing. Unlike the complex circuits discused in the forthcoming preamp survey, these are very simple 4- to 6-transistor circuits. It is unclear to me that these very simple discrete op-amps offer improved performance in comparison with high-quality integrated op-amps. Any advantage of a discrete signal path may be lost when the signal passes through the Dolby encoder and decoder, which are fully integrated.

Sonically the unit lost a small amount of detail in comparison with the CT-93, but I was comparing a brand-new CT-93 against a year-old TD4800. Better results might have been achieved if bias and level adjustments had been carried out with a more accurate level meter than the bar display on the TD4800. My use of Maxell XLII-S tape could also account for some of the difference. The fixed equalization adjustments on the TD4800 were set for TDK SA. Better results will be achieved with the TDK tape, but I would expect the formulation for this tape to evolve over time. Any new formulation would not perform optimally

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A 31" TV Monitor/Receiver with Audio Facilities

By Peter Aczel
Editor and Publisher

Here's a good example of the integration of audio and video in home entertainment systems using audiophile-quality components.

I promised to review at least one video product with a strong audio tie-in in every issue, and this one certainly meets that qualification.

Proton VT-331

Proton Corporation, 5630 Cerritos Avenue, Cypress, CA 90630. VT-331 Color TV Monitor/Receiver, $2200.00. Tested sample on loan from manufacturer.

This unit is designed as a TV module to be inserted into a stereo system, although it *will* produce puny sounds on its own, out of the built-in pair of small speakers and (so-called) 10-watt-per-channel amplifier. Proton, as a maker of separate tuners, amplifiers, CD players, etc., is well aware of what makes a good separate video component. The VT-331 has all the features required to interface with any stereo system without the slightest hassle.

I used the VT-331 for several months with the variable audio output going directly into a 200/200-watt stereo power amplifier connected to a pair of high-quality speakers. That way I could use the remote control for all audio and video functions, just as if I had been operating an ordinary TV set, and obtain the kind of sound quality I'm used to as long as the program had a decent sound track. I was quite happy with this setup as my main TV, except that I was a bit shocked how much smaller its 31" screen looked than that of the 32" Toshiba FST tube I had been watching before. Although the screen area is only about 6% smaller, subjectively it seemed more like 25%. (Some yo-yo will now say, "You see? He's admitting that subjective reviewing is the way to go!" No, yo-yos, 6% is correct; 25% is an incorrect subjective impression or optical illusion, not a deeper truth.) Anyway, the line-level stage of the audio section gave me no audible problems.

Other audio features of the VT-331 include fixed line-level outputs; genuine stereo from MTS stereo broadcasts (as well as videotapes and laser videodiscs, of course); an "expansion" effect that makes the stereo image appear larger (more useful with the built-in speakers than an external stereo system); a "pseudo stereo" effect to be used with mono sources (not if you ask me, though); and—most interesting of all—the Aphex Aural Exciter™, a signal processor with a studio pedigree, discussed at some length by Stanley Lipshitz in our "Seminar 1989" (see Part II in Issue No. 14, page 50). It's a kind of mid- and high-frequency enhancer, gimmicky but occasionally effective on pop-type material to obtain added "presence." Not for the purist but certainly a bonus feature on a color TV.

And that's not all, as they say in those special-offer commercials. The VT-331 also has a microphone input (for ordinary dynamic microphones only) with a separate mixer control that allows voice-overs, balancing the speech level against the sound track level, *karaoke* singing, and other neat tricks. That front panel is almost a miniconsole, so to speak. One must hand it to Proton for having produced an audio consumer's kind of TV, although the high-end audio snob will undoubtedly look upon some of these "bells and whistles" condescendingly.

Of course, even an audio consumer, high-end or not, wants more from his TV than lots of audio features. What about the video performance? As before, I used Joe Kane's laser videodisc *A Video Standard* (on the Reference Recordings label) as my source of test patterns. Black level retention—holding black at black, independently of picture content—was very acceptable but not outstanding (as is typically the case, except on professional monitors). The highest contrast level obtainable was not as contrasty as on some monitors, but it was more than adequate and was well within the peak linear capability of the set. Color performance via the S-video input was good; I think some would call it very good, but I wouldn't call it outstanding. The factory settings of Color and Tint (i.e., hue) weren't exactly the most correct—not that it matters a great deal when everything is controllable and programmable right on the screen, but the 32" Toshiba, for example, had been perfect in this respect, whereas the Reset command on the Proton gave me default settings that I could improve on. (In general, I consider the 32" Toshiba FST tube to be more nearly state-of-the-art in color performance; it delivers a more vivid, snappy color picture, although the Proton is no slouch in that department and is much more of a quality product as a total package than the Toshiba set I tested.) The geometry, on the other hand, appeared to be perfect on the VT-331; the checkerboard patterns showed no distortion. Convergence was also unexceptionable.

I haven't gone into the question of video bandwidth and horizontal resolution because under domestic (i.e., not laboratory) conditions it's entirely academic. Although 600 lines of resolution are claimed for the Proton (700 lines for the Toshiba), my very high-resolution videodisc player is specced at 440 lines (which I don't for a moment believe), and 336 lines are the theoretical best for NTSC broadcasts. The SMPTE resolution pattern indicates that the Proton is certainly up to the latter spec; more than that I'm hesitant to say. (The test pattern looked a little a better on the Toshiba than on the Proton; neither was flawless). Maybe a quote from Joe Kane's instruction booklet would be appropriate here: "Numbers being quoted for horizontal resolution are as inconsistent as amplifier power specifications in the early 1960's."

One of the best features of the VT-331 is the remote control. Angled like a dentist's mirror, curved to fit the hand, and uncluttered in its layout of buttons, it is ergonomically best I've ever used. Combined with the intelligent screen displays and menus, it helps to make channel hopping and video jockeying with the VT-331 a pleasure. My opinion of this monitor/receiver, on balance, is that its careful engineering, quality construction, convenience of use, and easy interface with typical audio component systems far outweigh whatever minor reservations I may have about its ultimate video performance, which is certainly quite good by most standards. The Proton VT-331 is definitely a worthy addition to a high-performance home stereo system. 0

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D/A Processors

(continued from page 38)

whistles, the comparably performing Digital Link II is a much better value, minor warts and all. Another alternative is to wait for the latest-and-greatest PS Audio digital processor, called—how did you guess?—the UltraLink. Announced at the Winter CES, this $1995.00 unit is designed around the very costly UltraAnalog 20-bit DAC, which uses hybrid circuit technology and is found only in the latest high-end processors selling at incomparably higher prices. PS Audio claims -95 dB THD + N with a full-scale input at 1 kHz, a better spec than anything I've measured so far (even including my delta-sigma explorations). Other claims are equally impressive. Although far from cheap, the UltraLink sounds like a bargain to me if it can really do all that. There may be a review of it in the very next issue. •

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Cassette Decks

(continued from page 41)

with the TD4800, and the original formulation would become unavailable when the new formulation was introduced. The use of a tape brand for which the machine is not optimally adjusted gives a better indication of long-term performance. I confirmed that Dolby S tapes are portable at least between state-of-the-art machines. Tapes made on the CT-93 sounded great on the TD4800, and the reverse condition also held.

Harman/Kardon has a long tradition of producing first-class cassette decks. They produced one of the first cassette decks, if not *the* first, capable of high-fidelity reproduction. Old Harman/Kardon cassette decks were a joy to use. I hope the major ergonomic problems of the TD4800 are just an aberration. •

Special Report

The 91st Audio Engineering Society Convention; or, The Invasion of the Credibility Snatchers

By Peter Aczel
Editor and Publisher

"Audio Fact & Fantasy: Reckoning with the Realities" was the theme of this convention, highlighted by the frightening attempt of golden-eared extraterrestrials to wrest AES credibility for their belief system.

The Audio Engineering Society is to audio what the NFL is to football. It's where the pros are. It's also the official custodian of professional standards and chronicler of professional activities in its field. I, myself, am a Life Member of the AES but very small fry next to such leading lights as Floyd Toole, Richard Small, John Eargle, Stanley Lipshitz, John Vanderkooy, David Clark, Ken Pohlmann, David Griesinger, Don Keele, Richard Greiner, Bart Locanthi, and so many others, most of whom participated in this particularly interesting convention, which took place in New York from October 4 to 8, 1991.

There were many technical sessions and many workshops, on a multitude of subjects, but the ones that interested me the most were those dealing with the chosen theme of the convention as quoted in the subhead above. One all-day session in particular, "Listening Tests" (chaired by David Clark), and one morning workshop, "New Cable Designs: Innovation or Consumer Fraud?" (chaired by Dan Dugan), are worth discussing here for their relevance to the editorial concerns of this publication.

The invasion by the golden-eared subjectivists, who generally stay away from the AES, was also targeted primarily toward these two gatherings. There was a sizable delegation from *Stereophile* (looking about as much at home as Baptists from Mississippi at a Park Avenue bar mitzvah), and quite a few other "antiverificationists" (definition below) surfaced during the discussions, to the dismay of the scientifically trained members. A more or less rational tone still prevailed, however, even at the invaded events—or so it seemed to me. What remains to be seen is whether any of the invaders will now try to use their participation in the convention as proof of the credibility of their ideas to the engineering community—which it isn't.

Polarity—when is it audible?

Polarity was the subject of three presentations at the session dealing with listening tests, two of them highly enlightening and one merely self-serving. Don Keele presented an elaborate mathematical model and measurement method for investigating polarity in complicated, real-world bandlimited systems, such as for example a multiway loudspeaker system. I think this very thorough study will prove to be particularly relevant to the design of DSP (digital signal processing) equipment for audio applications.

Professor Richard Greiner's outstanding paper on the audibility of acoustic polarity made me understand for the first time the complexities of the subject. For example, polarity inversion is more obviously audible when the loudspeaker used in the listening test has high second- and fourth-order harmonic distortion! Through a low-distortion speaker, only relatively simple asymmetrical waveforms (such as a sustained single note played on a trombone) are likely to sound different when the polarity is inverted, although the sound of certain instruments particularly rich in transients (piano, guitar) may also show a slight sensitivity to polarity inversion, even when the waveform has little asymmetry. Fascinating—and not nearly as black-and-white as Clark Johnsen tried to make it appear in his paper on absolute polarity, the main thrust of which was how right he has always been and how wrong everybody who disagrees with him has always been. Pretty lightweight stuff. I was particularly turned off by his "triple-blind" (huh?) listening test, for which he claims an additional level of blindness because in the first trial the listener doesn't know that a test is taking place. Actually it's a single-blind test because Johnsen, the test giver, knows all along what's happening and what answers he wants; furthermore, the test ignores the bias in favor of Different in a Same/Different test. (That's Tom Nousaine's favorite subject, which he discussed convincingly in a brief paper surveying published listening tests.)

The Harley debacle.

The climactic nonsense of the day was the paper (if one may dignify it with that name) by Robert Harley, the golden-eared Consulting Technical Editor of *Stereophile.* It consisted mainly of petulant objections to the idea of scientificverification in listening tests. Harley seriously believes that we should accept his exquisite sensibility at face value when he declares that amplifier A sounds much better than amplifier B. We shouldn't ask for verification that he can actually distinguish the two amplifiers by their sound when their identities are concealed. No double-blind tests, please, he says; no ABX-ing; no single-blind tests, either—they're all invalid because you can't subject music, which is an emotional experience, to crass objective test procedures. Hence my term "antiverificationist"—it's a whole new harassed audio philosophy.

It's not my intention to repeat here what I've written so many times before on this subject; a significant portion of Issue No. 16, among others, focuses on exactly these matters; but I still want to remind "RH"—how I dislike that clubby, insiderish alphabet soup!—and other deep thinkers of his persuasion that *the amplifier is not the music,* any more than the wineglass is the wine or the window is the view. I have a very low tolerance for pretentious utterances like Harley's about the emotional content of music when the subject is a piece of dead hardware that serves as *the conduit for music.* To pursue the analogy, the issue is not which of these two glasses of La Tache 1971 makes me feel more euphoric but which of the two glasses is more suitable for determining the true color of the wine and capturing its characteristic bouquet. This shouldn't really have to be pointed out to serious, thinking practitioners.

There was a moment of piercing illumination when I asked Bob Harley a question from the floor. I wanted him to tell me—since he claims to hear these differences that I can't, that my associates can't, that our double-blind tests can't verify, and that he himself is not willing to be tested on because all objective tests are invalid—how I can be sure that he is being truthful? My question received a spontaneous round of applause from the AES regulars, but he launched into a pompous discourse about individuals with extraordinarily keen sensory perceptions, etc., instead of answering my question. I asked him once again, "How do I know you are being truthful?" His reply: "How do I know *you* are being truthful?" This must have been considered brilliant by his friends and supporters in the room because they started to clap like crazy, whistle, and stomp their feet. I still can't figure out whether RH believes that we can all hear these things but deny it just to give him a hard time, or whether he just meant to say, "so's your mama." At about the same time a New York City sound-studio type named Bob Katz, who had been sitting with the *Stereophile* group, stepped up to the floor microphone and opined that Bob Harley's paper was the most important of the entire convention. It was that kind of session.

Following this donnybrook there was another episode of poignant human comedy, but of a different, more subtle kind. Michael A. Gerzon, the Oxford mathematician responsible for Ambisonics, delivered a paper titled "Limitations of Double-Blind AB Listening Tests." It was a highly technical and thought-provoking argument in favor of certain procedural changes in conventional double-blind testing, but John Atkinson, the militantly subjectivist editor of *Stereophile* (and Bob Harley's puppeteer), obviously thought it was going to be a delicious put-down of the ABX faction by—at last!—a highly accredited academician. There was no preprint to be looked at, so John started to scribble notes furiously as soon as Michael Gerzon opened his mouth. About five minutes into the presentation, the latter remarked that of course double-blind conditions are a must for any kind of validity in a listening test. John abruptly stopped taking notes as if someone had pulled his plug. It was a moment to be savored.

Another interesting paper on listening test procedures was the one by Floyd Toole, presenting details of his latest binaural experiments. He described "a binaural (dummy head) recording system [that] has the potential of capturing and storing an entire listening experience, and reproducing it for comparison with others at a later time." In other words, accurate subjective listening evaluations don't necessarily have to take place in real time anymore. (No, John and Bob, that's not an endorsement of anecdotal comparisons from auditory memory.) I was especially interested in the unique audiometric-type headphone chosen for these experiments, designed by Etymotic Research and utilizing a flexible probe tube inserted deep into the ear canal with the aid of a soft plastic-foam earplug. The very idea makes me squirm and cover my ears, but the response of the device blows away—for once the term is apt—that of the best conventional headphones with their inevitable interaction with the pinna (the outer ear). Floyd Toole's work is always highly creative and relevant to the most pressing problems in audio, so you can expect further comments on this subject in our pages after we've done some deeper digging.

The cable workshop.

A number of observers expressed the opinion that Dan Dugan (who comes from the professional side of the audio community and has nothing to do with the audiophile market) had "rigged the jury" of the cable workshop which he chaired in order to make the high-end cable protagonists look ridiculous. Well, they did look ridiculous, but no, it wasn't Dan's fault nor Dave Clark's, who was more or less the second in command. There were cable manufacturers on the floor of the meeting and they never said a word, although audience participation was heavily solicited.

As far as equal representation of the cable cult is concerned—and that's what most of the bitching was about—imagine trying to assemble a panel consisting of, say, three M.Ds and three witch doctors. It's difficult to put together that kind of combination. Corey Greenberg, the gonzo junior editor representing the far left wing of *Stereophile* (the Sumo brand evokes "butt cheeks" to him), was the only full-fledged cable cultist on the dais, and he appeared to be somewhat intimidated by all the engineering brass in the room. (As a matter of fact, I can't understand why *Stereophile* wanted Bob Harley and Corey Greenberg to be their official spokesmen at the AES instead of John Atkinson and Tom Norton. The cannon fodder concept?)

Those who were unfamiliar with the ABX double-blind comparison method had a rather superficial and unconvincing introduction to it—just a very few quickie trials, not enough to be an object lesson and not nearly strict enough in protocol—and that constituted legitimate grounds for complaint. On the other hand, the inhospitable AES reception of ignorant/immature objections to scientific fact should have been expected by the complainers—*c'est la vie* in a sophisticated engineering environment. I still get slightly annoyed—although I should know better by now—when the untutored bozos of the audio world want 2+2=4 and 2+2=5 to have "equal time" in a public forum.

One interesting feature of the cable workshop was the discussion of the possibility of intervention in the marketplace by city, state, and/or national consumer protection agencies—in other words, the specter of prosecution for consumer fraud. That certain claims for audio cables are consumer fraud is unquestionable; only the esoteric nature of the market has kept the authorities out of it so far. That could change. After the workshop, as the panelists were leaving, Larry Archibald, the entrepreneur who publishes *Stereophile,* tried to convince panelist Wilfredo Lopez of the New York City Department of Consumer Affairs that the opinions of the AES on this subject are quite untypical and that the mainstream of the audio community just loves expensive cables. Larry appeared to be quite agitated. Maybe he thought Mr. Lopez would now want to take action against stores like Lyric HiFi in Manhattan for selling zillion-dollar silver cable to brainwashed innocents. (It was there that I once overheard their ace salesman say to a customer, "Well, of course, silver cable is twice as fast.")

Overall, the cable workshop pointed toward the same conclusions as I have presented in my cable articles. Surprise, surprise.

The low bit-rate situation.

An important seminar at the convention explored the hottest subject in audio today, namely low bit-rate coding for the purpose of reducing data capacity requirements. The digital compact cassette (DCC) about to be marketed by Philips is just one example of this new technology; others include the Sony mini-CD and digital audio broadcasting—all of them expected to be staple formats of the 1990s.

The seminar was chaired by Ken Pohlmann and included highly accredited participants like John Eargle, Louis Fielder (of Dolby Laboratories), and Bart Locanthi (heard on tape only). The subject is much too broad and deep to be discussed here even glancingly, but this was my main take-home impression: there's nothing wrong with the principle of leaving out data we can't possibly hear, but there can be everything wrong with the way it's done. This is a crucial period of standardization. What we do now is for keeps. We can't afford to have a bad standard for low bit-rate coding because the future of consumer audio is at stake. (Remember—we could have had the superb Crosby system of FM stereo broadcasting, but we ended up with a noise-prone compromise.) What private companies like Philips and Sony do with their proprietary products we have little control over, but at least the digital audio broadcasting standard should be the best possible. It could be that years from now there will be magnetic and laser-scanned media of much higher data storage capacity than what we have today, so that we may end up regretting at least some of this now-and-forever formatting, but the low bit-rate approach is a certainty for digital audio broadcasting. Let's not blow that one. •

A Hasty Postscript: "There You Go Again!"

Long after the above was written and shortly before press time, the January 1992 issue of *Stereophile* came in the mail. John Atkinson quotes my dialogue with Bob Harley verbatim, ostensibly to show how wrongheaded I was but in reality to have a vehicle for a sneaky footnote, which would have appeared too poisonous without such a framework. It suggests that I somehow misappropriated the *Stereophile* mailing list for our recent promotional mailing, whereas the truth is that the mailing went to a commercially available list that had been routinely rented in the open market. Nasty!

Then JA goes on to call me, David Clark, and Dan Dugan "dinosaurs" on one side of these issues, just like Peter Belt, George Tice (!), and Jimmy Hughes on the other side; *Stereophile,* however, occupies a middle ground, he says, along with—get this!—Stanley Lipshitz and Floyd Toole.

Well, I've got news for you, John. Stanley Lipshitz and Floyd Toole have just as little respect for your position on double-blind testing as I do and don't see you as a moderate. Go ahead, ask them. And another thing—just because you and Bob Harley go around wearing Robert Pirsig and Michael Polanyi T-shirts (figuratively speaking) you aren't impressing anyone who knows something about the philosophy of science. Pirsig is an entertaining didactic/literary lightweight who doesn't take himself nearly as seriously as you take him; as an argument-clinching philosophical reference he is a joke. Polanyi is more substantial, but his romantic antirationalism and glorification of intuition in scientific discovery have not been accepted by the vast majority of modern scientists, and in any case the uproar about double-blind listening tests has nothing to do with scientific discovery—it's simply a question of level matching and no peeking. Why all the pretentious verbiage?

The Pirsig/Polanyi incantation is only one of the minor features of Bob Harley's lead article in the same issue, titled "Audio McCarthyism" and annotated with footnotes by JA. Their main point: Dan Dugan's cable workshop used McCarthy-like tactics of accusation and intimidation against those who claim to hear differences between loudspeaker cables. My impression is that with this embarrassing editorial tantrum the Atkinson/Harley faction has crossed over from muddleheaded defensiveness to freaked-out, paranoid raving. Are you talking about audio, RH and JA or our political freedom? Just think for a moment. The essence of McCarthyism was accusation without proof. The basic rap against you subjectivists is that *you* make assertions without proof, that you resist the process of verification and accept anecdotal information as fact. Isn't the McCarthy analogy therefore a bit on the brain-damaged side? •

More about the AES Convention

The Blind Misleading the Blind

By Jeff Corey, Ph.D.
Professor of Psychology
C. W. Post College of Long Island University

Some of this covers the same ground as the preceding article but from the point of view of an academic in experimental psychology.

Dan Dugan asked me to attend the October AES convention in New York and participate in a workshop titled "New Cable Designs: Innovation or Consumer Fraud?" in my capacity as an experimental psychologist and member of the New York Area Skeptics. It was my first introduction to the so-called Great Debate between the "subjectivists" and "objectivists" on the question of assessing the quality of sound reproduction. I haven't seen that kind of "debate" since the behaviorist-humanist conflict in psychology back in the 1960s.

My contribution was to explain the necessity of double-blind procedures in any subjective testing situation. I focused on the problems involved in subjective judgments and the need for double-blind procedures to control subject and experimenter bias (the Rosenthal effect). Experimenter and observer bias need not be evidence of fraud; rather, they are common behavioral phenomena. In addition, I was asked to explain why some people might continue to insist on hearing differences when no evidence supported their existence. I concluded by relating the tale of P.T.Barnum and his "Feegee Mermaid." Everyone at the workshop was relatively polite, which did not prepare me for the later level of discourse.

After the workshop, I was approached by a number of people who took issue with my point of view. The first identified himself as a certified clinical psychologist and audiophile. He insisted that I must listen to his sound system and judge for myself. I replied that I would do so, but only in a double-blind test. After we both persisted in our different views, he stalked away crying, "And you call yourself a scientist?" Another gentleman introduced himself with, "I attended Harvard and have a degree in physics. I invented the triple-blind method." *[This was Clark Johnsen. See also the preceding article.—Ed.]* Then he gave me a preprint. I avidly scanned the preprint because the only triple-blind method I ever heard of was the old joke about doing a double-blind experiment and then losing your data.

As it became apparent upon reading the paper, this triple-blind procedure was a joke, too. The new twist turned out to be that the subjects were not told that they were being tested. They were presented with recorded material in "reverse polarity" and then, after a selection played for a while, "the operator (so as not to cause undue suspicion) winked at the subject and spoke words to the effect, 'We'll do that again.'" Then the operator switched the wires to the correct polarity and "stayed out of sight to fufill the criterion of a blind (unseen) operator." After the test was over, the operator returned to the listening room, gave the subject a response sheet and asked, "Did you hear a difference?"

This exemplifies one of my favorite experimental psychology exam questions. Given an example of a hopelessly flawed study, how many flaws can the student find? An adequate scholar would list:

1. No informed consent was obtained.
2. Reverse polarity was always presented first. A confounding order effect is likely, as shown in a later paper by Tom Nousaine.
3. The operator always knows the condition being presented. Verbal or nonverbal cues may bias the subject against the first presentation (with the operator present). The next cue that something different (maybe even better) is coming up is given away by the wink. Finally, being away during the second presentation does not count. The key point is that the potentially biased experimenter was present during the collection of the data.
4. There is no guarantee that the procedure of reversing polarity did not affect other salient features of the presentation, such as the exact decibel level, which can affect subjective ratings (Nousaine's paper again).

While polarity may have a real effect, this study did not demonstrate anything warranting the author's conclusions. The triple-blind design is a hoax. When I questioned the presenter later, he replied that he was a "great actor" and could disguise his bias from the subject. Nod, nod, wink, wink. Or as the Pythons put it, "A nod is as good as a wink to a blind wombat."

The next day, I heard Robert Harley, of *Stereophile* magazine, attack the "objectivists." He objected to well-controlled experiments in listening on many grounds: they put pressure on the subject, components are switched too rapidly, and the scientists who run these tests are biased toward the conclusion that no differences exist (all the more reason to employ a double-blind procedure). Among the more ludicrous arguments was that if double-blind testing did not reveal differences which he could hear subjectively, then there is something wrong with blind testing. This reminds me of discussions with New Age (rhymes with sewage) believers in Crystal Power or ESP who, when they are faced with the fact that no scientifically sound, well-controlled, replicable data exist to support their claims, say, "Well, man, you skeptical scientists give off bad vibes that inhibit the Power—that's why, man, like be holistic or something." Say, Mr. Barnum, do you think these yokels would go for New Age Crystal Power Cables, filled with Thousands of Magical Silicon Dioxide All-Natural Crystals?

Let's conduct a thought experiment like good quantum mechanics. Suppose we were to persuade some audiophiles to take a subjective test. Let them relax with their favorite selections played over a decent system. Pretend to plug in some well-known expensive cables and alternate them with some cheap cables whenever they wish. Go back and forth as many times as they wish and ask them to rate the performance on such key scales as "liquidity of midrange" or "graininess of highs," or just ask for their preference. Take your time; be holistic. However, make sure that all selections are played through the same cables and the switching is done with cables that are not connected to the amp or speakers.

The experimenter who apparently switches the cables and collects the data would have to be ignorant about the real purpose of the study. The results might show a strong placebo effect. Easy enough to do, but it would suffer from a certain ethical problem—misinforming the subject. That's what makes it a thought experiment.

Well, I learned one thing from the AES convention. Next time I buy equipment, it won't be a $5995 tube amplifier and I won't use $100-per-foot silver cable to hook up my speakers. •

Reprinted by Popular Demand: Our 1978 Classic

This almost 14-year old cartoon by Dick Calderhead was probably the most talked-about in our history. It offended about three and a half bluenoses out of many thousands of readers at the time, but it happens to be as apropos today as it was then—witness the preceding two reports. The absence of a Tom Aczel cartoon—due to circumstances beyond his and our control—seems to be a good opportunity to respond to the requests of charter subscribers who feel that the newer generation of readers should be exposed to this classic commentary on the audiophile scene.


IN YOUR EAR
"AT LAST I HEAR THE FRONT-TO-BACK DEPTH I ALWAYS WANTED TO HEAR."
CALDERHEAD

The Wire and Cable Scene: Facts, Fictions, and Frauds Part III

By Peter Aczel
Editor and Publisher

This part deals mainly with the ridiculously simple subject of line-level interconnects, after tidying up a few loose ends left over from the speaker cable article in the last issue.

If, in the last installment, I got across to the reader the idea that a speaker cable is nothing but an RLC circuit and therefore can be expected to behave like one within the network which it forms with the amplifier and the loudspeaker, then my mission was accomplished. I didn't cover a number of peripheral subjects, however, that need to be brought up.

Connecting the speaker cable to the terminals.

If you don't plan to swap amplifiers, speakers, and/or cables in the foreseeable future, and if you're not in the habit of unplugging them and moving them around, I suggest that you use no hardware at all for connecting the ends of your speaker cable to the amplifier and speaker terminals. Just put the stripped ends through the holes in the terminal posts (if the gauge of the wire is too big for the holes you've already been had by the cable cultists) and tighten the screw-down sleeve of each terminal as hard as you can, preferably with a hex-nut driver. Such a connection is the kind least likely to loosen and give you trouble. Remember: a loose connection is much more likely to give you "bad sound" than any allegedly low-fi cable in existence.

Oh, yes, some people swear by "contact conditioners" like Cramolin or Tweek; I never found it necessary to use them, but if they make you feel better, go ahead. Like chicken soup, they can't hurt, and they don't cost all that much.

I realize, of course, that not all amplifier and speaker terminals are of the binding post type; some, for example, will accept only banana plugs, in which case the above advice is inapplicable. If you need hardware to terminate your cables, that's the occasion to spend a little more money than some skeptics may be inclined to. Get the best-made plugs you can find. The minimal, cheesy kind will give you endless and unnecessary trouble—separating from the cable, not fitting into the jack tightly, constantly oxidizing, etc.

The RFI issue.

Radio Frequency Interference (RFI) is a complex and dramatic phenomenon in which the speaker cable may conceivably play a role, though hardly ever the lead. It has happened to me a couple of times that I heard "Breaker two! Breaker two!" coming out of my left or right speaker after some snap, crackle, and pop, but every time it was traceable to something further upstream in the system than the speaker cable. Of course, some very temporary snap, crackle, pop, or hash could also be RFI, without a good buddy's voice materializing before your very ears, and it could even be due to the speaker cable under exceptional circumstances, but one thing is certain: it can't possibly have anything to do with the metallurgy of the cable—whether it's made of silver or oxygen-free copper, or with perfect crystal structure, and all that jazz. In the unlikely event that the speaker cable is the culprit, the cause is probably the orientation of the cable, in the antenna sense, and only as a distant second possibility the construction of the cable.

I'm in the process of trying to measure RF pickup by speaker cables of various types of construction and will report on my findings if and when I obtain some meaningful results. Quite frankly, it isn't a top-priority project on my calendar, but I'm mildly curious.

And now for interconnects...

A cable is a cable—or, more precisely, a two-conductor cable is a two-conductor cable—and a line-level interconnect can be represented by the same lumped-element equivalent circuit as a speaker cable, shown here again in Figure 1. (See also the discussion in Issue No. 16.)

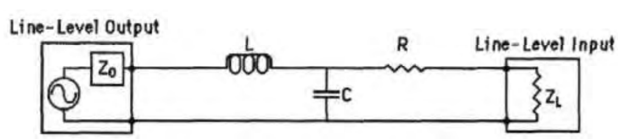


Figure 1: Equivalent circuit of an audio cable connecting a line-level signal source to a line-level input, with lumped circuit elements.

As in Part II of this series in the last issue, I used the RLC values measured by Martin Colloms, in this case for a June 1990 article in *Hi-Fi News & Record Review*, where he reported on 54 different interconnects of 23 different makes, with all parameters carefully tabulated, including the "ambience" and "pace" ratings of each cable. As I stated in the speaker cable article, I trust the English pundit's RLC measurements as completely as I disdain his irresponsible subjective pronouncements. (That cable's got rhythm, mate.) He has saved me a lot of routine, boring laboratory work, and I appreciate that.

In any event, the RLC values of interconnects are of considerably smaller consequence than those of speaker cables; my frequency response simulations—once again with the aid of Micro-Cap II for the Macintosh—revealed mostly insignificant departures from perfectly flat, regardless of the cable models, lengths, source impedances, and load impedances I plugged into the program. I had to search for extreme cases with improbable parameters to come up with something dramatic. Here's one.

The PRO-10MC Sonata Series preamplifier from B&K Components is a very nice $698 unit whose 20 dB line amplifier has a normally low output impedance, *but*—a Direct Bypass switch on the front panel can disconnect the line stage and allow the Volume and Balance controls to drive the output cables directly. This completely unnecessary Passive mode, put in strictly for marketing reasons to massage tweako audiophile preconceptions, can result in an output impedance as high as 50,000 ohms. Now let's suppose, since we're dealing with at least borderline tweako propensities, that the salesman managed to sell the customer a pair of 2-meter AudioQuest Turquoise cables, for an additional $60. Let's further suppose that this particular audiophile's power amplifier has an input impedance of 100,000 ohms, which is quite common. And, of course, our man will certainly switch to the Passive mode from time to time because it's "cleaner," right? Now we've got something. The response of this setup is shown in Figure 2. With a top-end rolloff to -3 dB at 10 kHz and -7 dB at 20 kHz, I'm sure that golden ears will find the highs very smooth, not at all grainy, remarkably free from transistory hardness. Gotta have good cables, man, especially when in the mercilessly revealing straight-wire bypass mode, you know...

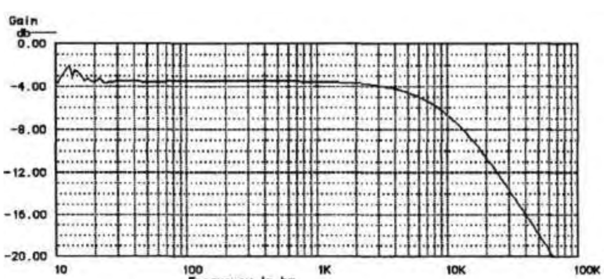


Figure 2: Response with 2 meters of AudioQuest Turquoise cable driven from B&K PRO-10MC Sonata Series preamp in the bypass mode into 100 k .

Let's now try a real golden-ear tweako special. The hypothetical owner of this equipment wouldn't touch a transistor preamp with a ten-foot pole because they all sound terrible, even the most expensive ones—don't they?—so he opts for the David Berning TF-12 tube preamplifier ($3245.00 minimum, more with options). The output impedance of this unit is 3000 ohms. He also buys 10-meter lengths of Monster Cable Sigma interconnects (approximately $7500 the pair) because he can afford them and wants to hide his mono power amplifiers behind his speakers, away from the control center. Those power amps also have an input impedance of 100,000 ohms. The resulting response is shown in Figure 3.

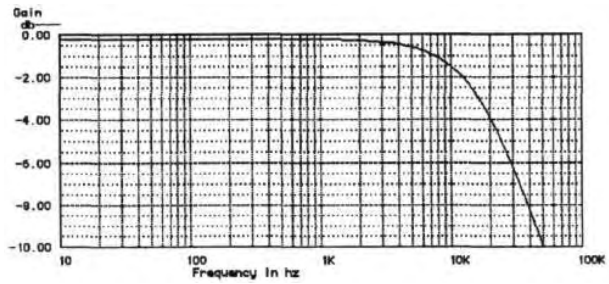


Figure 3: Response with 10 meters of Monster Cable Sigma interconnect driven from the David Berning TF-12 tube preamp into 100 k .

Well, what do you think the golden ears will make of -1.3 dB at 10 kHz and -3.8 dB at 20 kHz? Quite Wadia-ish, isn't it? Those Sigma cables sure sound sweet, unaggressive, and musical, don't they? Very different from your run-of-the-mill cables in a flat system, eh?

Seriously, though, these are examples I had to reach for; they're not what you're likely to find in the real world out there. Most line-level outputs in today's audio universe are in the 25 to 600 ohm impedance range, and not even the most capacitive interconnects will roll off the highs at domestic lengths with that kind of drive. It's basically a nonissue. With a few bizarre exceptions, interconnects have no effect on frequency response. Frequency response is by far the most audible parameter of sound-reproducing equipment—it could be argued it's the only one under normal conditions—and what else can an interconnect introduce besides frequency response changes? Nonlinear distortions? Hardly—unless there's something wrong with the plugs or the integrity of the conductors. A bad connection is in effect a diode.

Well then, what about construction quality?

If you want to pursue that subject, Martin Colloms's evaluation of the "Build Quality" of his 54 interconnects makes rather poignant reading. His ratings are good, good+, very good, and excellent. Among the 9 he rates "excellent" are the two cheapest in the survey, by Audio Technica and Sony. The lowest-rated 8 (merely "good") include the two insanely expensive Kimber pure-silver cables that he rates first and second sonically. This confirms the frequently experienced "wait a minute, we're having cable problems, we've got to check our connections" in ultrahigh-end demonstrations—loose plugs, bad grounds, broken joints, frayed shields, etc.—which those of us who use the cables that came with the Sony CD player hardly ever have to think about. Ironic, to say the least.

As I said before, I'm very much a believer in high-quality plugs. Nothing is more annoying than to have no signal, or a crackling signal, and then discover that a cheap RCA-type plug isn't gripping or has become corroded. My favorite RCA-type plug is still the tried-and-true Tiffany, for two reasons: (1) its construction assures a strong, reliable grip without requiring a major struggle when disconnecting; (2) it connects the ground *before* the hot pin, and disconnects it *after* the hot pin, when used in combination with a Tiffany female connector. The less costly plugs, however, have been getting better lately; a heavy-duty plastic body and gold-plated contact surfaces are the rule now even on the plugs that terminate the interconnects supplied free of charge with upper-medium-priced components. The best, most reliable cable connectors by far are the Cannon XLR-type professional three-pin plugs and sockets for balanced-line operation—and that brings us to an important point.

The balanced line.

As I've stated in the past, the balanced line is the obvious, natural way to go in audio design, whereas using the ground as the signal return path is something of a compromise, justifiable in terms of the audio industry's evolution but certainly not a class act. Isolation from all the garbage that can creep into the signal through the ground is a very desirable thing. Completely balanced operation of a particular design from input to output isn't always possible or practical, but more and more high-end components can now be interconnected at line level with balanced lines, a welcome development. (Yes, it requires additional active stages, but with today's superior active devices I don't consider that to be a drawback.) So here's the point I wanted to make: where are all the exotic high-end cables for the new high-end components with balanced inputs/outputs? There are surprisingly few, a tiny fraction of the tweako cable universe. It seems that the oxygen-free-kryptonite-litz lobby is being a little timid about this territory that used to be the private domain of studio professionals. That's really wonderful because it means that most balanced-line installations will use practical, cost-effective cables made by Belden, Canare, etc., without occasioning high-end-hypochondriac self-doubts. The situation won't last, of course; you can't keep the snake-oil vendors out of a growing market.

How to ground the shield of a balanced line is a mini-discipline I won't go into here because the information is available from many sources, including the makers of the components with balanced inputs/outputs, but it must be done properly. A balanced line with a correctly grounded shield permits very long runs of cable at line level while providing maximum protection against hum and RFI, both of which can be more and more of a problem the further upstream you go in the chain of reproduction. The long line-level connection makes it possible to place the power amplifiers) close to the loudspeakers so that short speaker cables can be used. That's my preferred configuration, but please don't quote me as saying that conventional single-ended interconnects with RCA-type plugs are no good. In most cases they're inevitable.

Back to the sound...

So, if we assume (1) no frequency-response anomalies due to weird source/load impedances, (2) quality construction without contact/conduction problems, and (3) adequate shielding and grounding to prevent hum and RFI, how could comparable lengths of interconnects sound different? Nohow. There exists no mechanism whereby an audible difference could occur. The metallurgical/geometrical/dielectric arguments are either total nonsense or at least irrelevant to audio frequencies (as distinct from the megahertz and gigahertz bands). Try to find a physicist, electrical engineer, or psychoacoustician—with a university graduate degree and no connection with an audio cable company—who will endorse those arguments or agree with Martin Colloms, Dick Olsher and company on their flights of fantasy about the sound of interconnects. The whole scene is a farce.

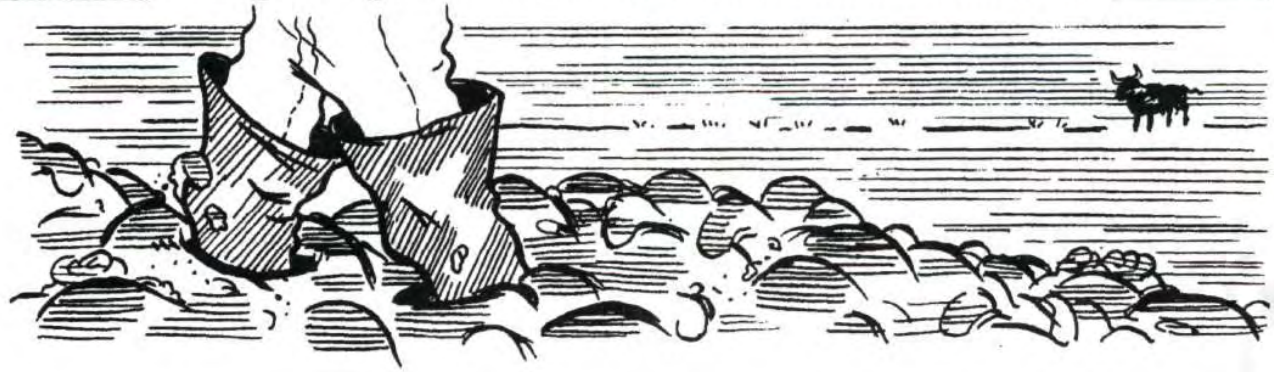
If you want to do a simple experiment to confirm the above, try this: Borrow a pair of insanely expensive, politically correct interconnects, silver or suchlike, from a high-end dealer. (Some of them will let you, especially with 1-meter or 2-meter lengths.) Then get a pair of inexpensive but decently made cables of the same length. Listen to both in your system (which I'm assuming is fairly normal in terms of output and input impedances). If you think you hear differences, try to zero in on them and memorize them.

Now have a friend toss a quarter or a half-dollar 16 times while you are out of the room and have him make a secret written record of the heads/tails sequence. Heads will stand for the high-end cables, tails for the others. Then come back and have your friend plug in the proper cables in that sequence while you listen to the same short piece of music over and over again. You must be blindfolded, or your friend must be in an invisible location, and there must be no talking, no throat-clearing, no muttering, no communication whatsoever. Write down for each of the 16 trials which cable you think is plugged in. Don't score the answers until the whole test is over. If you get 8 right, it means nothing; you might as well have been guessing wildly. If you get 9, 10, or even 11 right, it means little more; you were probably just lucky. If you get 12 or 13 right, I'll admit that there may very well be a real difference. If you get 14 or more right, there's surely a difference. But don't try to cheat; if you discover some clue other than the sound of the music—such as, for example, a tiny hum when one of the cables is plugged in—the test is invalid.

Without such a test—subjective but at the same time allowing objective verification—any discussion of the sound of interconnects is totally lacking in credibility. Elsewhere in this issue, the logical, psychological, and political aspects of listening tests are discussed at some length. •

Hip Boots

Wading through the Mire of Misinformation in the Audio Press



Editor's Note: *This column is in danger of losing its special identity in this issue because parts of the AES Convention reports are also typical "Hip Boots" material. That, however, is a nonrecurrent situation; normally the choicest audio absurdities appearing in print are channeled through here rather than the more serious editorial sections. So, for the sake of continuity, here goes...*

Shortly after the publication of Issue No. 16, I received a telephone call from Robert E. Greene, the UCLA mathematician who appears to be the most levelheaded of *The Absolute Sound's* mostly tweako equipment reviewers. The call was about the "Hip Boots" treatment accorded to the Tice TPT Clock in that issue and the reference to the November/December 1990 *TAS* articles endorsing the clock. REG basically wanted to distance himself from Michael Fremer's and Frank Doris's self-indulgent paeans to the crackpot Tice device, pointing out that his comments, which followed theirs, did not constitute an endorsement of the product. I reread all three articles, and it's true; he went no further than to take a let's-wait-and-see position and concede borderline plausibility to the Tice claims even if they should eventually prove to be false. I think that's bad enough, but I promised to publish this emendation. (All the while I was thinking, "What's a nice academic like you doing in a magazine like that?")

HP's editorial reply in *The Absolute Sound.*

Unlike Robert E. Green, Harry Pearson doesn't have to worry about losing face. He lost it long ago in the eyes of those of us who respect science and causality, while his untutored fan club couldn't care less. Why bother, then, with yet another of his insults to our collective intelligence? Mainly because this one has to do with a concept that all audiophiles need to understand very clearly but often don't, and secondly because this time HP (all that alphabet soup comes from *his* kitchen, by the way, not mine) has crossed the frontier between astrology and pathology, and a warning shot is in order.

A San Diego reader of *TAS* points out in a letter to the editor (Issue 74, November/December 1991) that in digital audio a loss of quality when making a digital-to-digital, bit-for-bit copy of a recording is utterly impossible because the copy will contain exactly the same codes, digit for digit, as the original. Apparently there was some tweako suggestion to the contrary in an earlier issue. This highly intelligent reader tries to explain that a digital copy "is either corrected 100 percent or it falls on its face completely: we get a loud clicking noise or muting. There is no such thing as a 'slight deterioration' or any other analog-like change."

Editor HP's reply: "You silly twit. You presumably have ears. Use them." And later: "...why don't you get out in the world and talk to those engineers who know that there are losses in digital copies... Talk about living in an ivory tower." And so forth. He even classifies the letter writer as a flat-earth dogmatist and himself as a round-earth realist!

That's what I mean by pathology. This is sick stuff. It seems that HP is suffering from the Emperor Jones Syndrome: to a few ignorant savages of audio he is emperor, ergo any word that issues from his mouth is by definition some kind of imperial truth. He said it, so it must be true. How does this man who shows no evidence of having had one semester of scientific education presume to contradict elementary scientific facts? Is he totally cynical and irresponsible or is he totally unaware of his own shortcomings? (Mind you, I think he is in a very general sense a gifted journalist. He should just stay the hell out of physics, mathematics, electrical engineering, electroacoustics, and experimental psychology.) And where are "those engineers" who talk such drivel? Where are their credentials? Do they have EE degrees? Or maybe that opinionated 19-year old studio gofer in the Metallica T-shirt, the one who cleans the tape heads, is an "engineer"?

Let's get this straight once and for all. An analog sound recording contains something like a map or diagram of the signal. A digital sound recording contains data, just a bunch of numbers (safeguarded by means of redundancy and error correction). That's a big difference. Subtle distortions in the chain of digital recording can occur in the A/D and D/A conversion stages—the conversion into numbers and back again—but the numbers themselves cannot become subtly bruised or altered. That's the whole point of the digital system of storage. Think of a $100 bill. You can draw a mustache on Benjamin Franklin's face; you can put a big tear into the bill; you can throw it into the washing machine in the pocket of your Levis—and it's still a $100 bill. You can change it at the bank; the teller won't give you only nine $10 bills for it because it's in bad shape. You'd have to shred it into tiny pieces or burn it to make it unnegotiable as $100. The numbers in a digital recording are exactly like that—an either-or situation, 100% recognizable or a total loss. Cut this paragraph out, HP, and paste it into your Mouseketeer hat.

Vance Dickason in his latest loudspeaker book.

"To knock a thing down, especially if it is cocked at an arrogant angle, is a deep delight to the blood," the great George Santayana once wrote, and of course that's part of the fun of "Hip Boots." This particular item is no fun, however; I wish there were no misinformation here to be pointed out.

I've always thought that *The Loudspeaker Design Cookbook* by Vance Dickason (published by Audio Amateur Press, Peterborough, NH) was a basically competent and reliable guide for the technically inclined loudspeaker enthusiast and amateur constructor. It went through three editions with only minor scientific inaccuracies in its pages; it sold 20,000 copies worldwide; it even gained acceptance in universities and technical schools as a textbook. That last development now becomes somewhat ominous because the Fourth Edition of the book, published in October 1991, shows a steep decline in scientific accuracy and accountability. Remember, we're talking about a sourcebook here, not the scribblings of some tweako audio journalist.

This new edition is twice as fat as the previous one and contains a great deal of new material. That would be wonderful if all the new material were accurate and instructive, but it isn't. The part that disturbs me particularly is the greatly expanded chapter on crossover networks. Filter theory is a fairly abstruse aspect of loudspeaker system design and Vance Dickason stands it on its head. I shudder to think that a whole generation of amateur loudspeaker techies will learn their filter basics from the LDC (clubby abbreviation for the book). This column is hardly the place for a detailed mathematical analysis of the misinformation in the crossover chapter, but a couple of examples are in order.

"Fourth-order Butterworth: We build fourth-order filters by cascading two second-order types. Since the Q of each second-order section of the fourth-order Butterworth is 0.841, the total Q is 0.707." This is complete nonsense. The two second-order sections have different Qs, neither of which is 0.841 (the correct numbers are 0.54 and 1.31); a fourth-order filter has no "total Q" as such; and you don't obtain a "total Q" by multiplying together two sectional Qs. Yes, 0.841 x 0.841 = 0.707, but it's totally irrelevant and inapplicable in this instance.

The fallacy of Q-times-Q-equals-total-Q pervades the entire chapter. About fourth-order Linkwitz-Riley crossovers it states: "Both second-order sections have a Q of 0.707, for a total Q of 0.49, which is why this filter is sometimes referred to as the squared Butterworth filter." Wrong, wrong, wrong, wrong. First of all, 0.707 squared is exactly 0.500, not 0.49, but that's irrelevant. Secondly, you can't multiply the Qs of the second-order sections to get a "total Q," as I already pointed out. Thirdly, only a second-order filter has a Q; a fourth-order filter doesn't—it has *two* Qs that don't get totaled. Fourthly, you have to look at the transfer functions to understand why it's called a squared Butterworth filter; it's not the Q that gets squared. Some sourcebook.

I'm not going to blame Vance Dickason alone for all this. His publisher should have had the manuscript reviewed by some authoritative academics, just as I had the above statements reviewed by Dr. David Rich, who happens to teach a filter course this year at the Polytechnic University in Brooklyn, New York. Maybe the advertisers who bought the three dozen or so advertising pages in the back of the book were impatient and had to see the LDC in print by a certain time—who knows. It's a pity because the book definitely fills a need and is worthwhile in many respects.

J. Gordon Holt on push-pull amps in *Stereophile*.

I have considerably more respect for J. Gordon Holt than for the present regime at *Stereophile*, but one of his increasingly rare reviews—in the October 1991 issue, critiquing the Boulder 500AE power amplifier, which I myself own—seems to indicate that the technical howlers of the Olsher/Harley brain trust have become contagious. As the subject is electronic circuitry, and in view of my basically cordial relationship with Gordon, I wanted David Rich to write up this one.

Dr. Rich writes: "In this review, Mr. Holt attempts to explain an important characteristic of push-pull amplification systems. In his discussion, Mr. Holt makes several very significant errors.

"In a push-pull amplifier the input signal and its complement are presented to two identical nonlinear amplification elements. The outputs of the two amplifiers are then subtracted one from the other. Ideally, when the nonlinearities are identical, the even-order harmonic distortion products at the output of the amplifier will be canceled. This property of push-pull amplifiers can be shown through the use of a power series expansion. (The Editor believes that presenting power series expansions in this column will result in the MEGO—My Eyes Glaze Over—effect, so no proof will be presented.) Mr. Holt states that this property will not hold for a class B amplifier. This is incorrect. In a class B amplifier, each amplification element conducts for only half the period of the input signal. This is a very significant nonlinearity but it is complementary, and even-order harmonics will be canceled. The principal disadvantage of a class B amplifiers is crossover distortion, which will result in higher levels of *odd-order* harmonics.

"The second significant error Mr. Holt makes is to state that a complementary emitter-follower push-pull stage does not exhibit cancellation of even-order harmonics. Mr. Holt does not seem to understand that, by definition, a complementary device provides the requisite inversions at its input and output. Vacuum tubes are not available in complementary form; therefore a phase-inversion circuit and a transformer are required to form a vacuum-tube push-pull amplifier. The availability of complementary transistors is a significant advantage of transistor amplifiers. Mr Holt draws exactly the opposite conclusion because of his misunderstanding of push-pull amplifiers.

"The properties I have outlined above are taught to almost every junior electrical-engineering student, yet nobody at *Stereophile* caught the errors made by Mr Holt before his work was published. Since nobody on the *Stereophile* editorial review staff was capable of catching these fundamental errors, it should come as no surprise that *Stereophile* magazine is riddled with errors when discussing advanced topics in electrical engineering, such as data conversion, serial data transmission, digital signal processing, and electromagnetic theory (recall the highly magnetic wire debacle). It is also interesting to note that Jeff Nelson, president of Boulder Amplifiers, did not choose to correct Mr Holt in his manufacturer's reply."

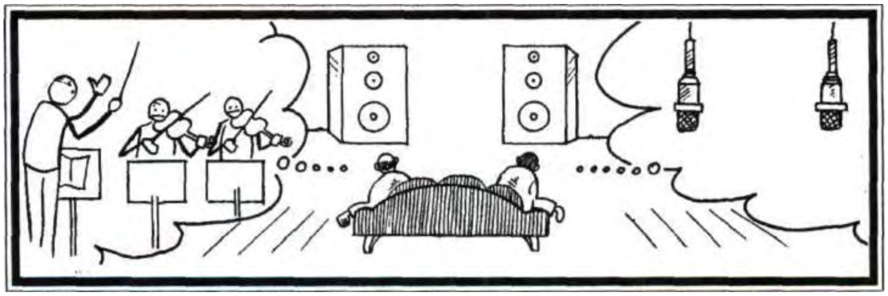
Also sprach David Rich. All I wish to add to his last paragraph is an ancient Hungarian proverb that translates, "You can't ride on two horses with one arse." The two divergently galloping horses of science and tweako subjectivism are too much for the editorial buttocks of any one audiophile magazine. •

Whatever Became of the Preamplifier Survey?

Yes, it was announced on the back cover of Issue No. 16 and, yes, it's coming. What happened was that David Rich made a bigger thing out of it than planned—that's his style—and it's still growing as this issue goes to press. It will definitely be published in Issue No. 18, and I can tell you right now that it will be about engineering excellence, reliability, and value for your dollar, where there exist big differences, rather than about the liquidity of the upper midrange and the airiness of the highs, where one preamp is as good as the next. You won't find preamp reviews like David Rich's anywhere else.

—Ed.

Recorded Music



This column is still looking for a professional music critic who is also knowledgeable about audio and isn't completely jaded when it comes to the standard repertory. Until such a rare bird comes to roost on our editorial perch (several—for classical, jazz, rock/pop, etc.—would be almost too good to be true), your Editor will have to muddle through on his own. Please tell us how you feel about the new format being tried out here, emphasizing broad horizontal coverage: one feature review followed by tabular capsule reviews, but lots of them. We can always go back to the old format if you liked it better.

Eliahu Inbal's Berlioz Cycle on Denon

By Peter Aczel
Editor and Publisher

It took Eliahu Inbal, the Jerusalem-born 55-year old international conductor, less than two years, from September 1987 to June 1989, to record with the Frankfurt Radio Symphony Orchestra seven of the major works of Berlioz for the Denon label. All seven were recorded in the Alte Oper in Frankfurt, with Yoshiharu Kawaguchi and Richard Hauck credited as producers and Detlev Kittler as recording engineer. Brtiel & Kjaer microphones and Nippon Columbia's digital-delay technique were used in all the recording sessions. As an integrated package from a single source with a single point of view, it's state-of-the-art Berlioz, but of course the art of music doesn't quite work that way.

Here's the list of albums with the Denon CD numbers:

Symphonie Fantastique, Op. 14	81757 3208 2
Harold en Italie, Op. 16	81757 3207 2
Requiem, Op. 5	81757 3205 2
Romio et Juliette, Op. 17	81757 3210 2
La damnation de Faust, Op. 24	81757 9200 2
Te Deum, Op. 22	81757 6142 2
L'enfance du Christ, Op. 25	81757 6863 2

The list is in the order of composition; the last Denon release was the *Faust*, which I received only very recently.

I'm a Berlioz enthusiast and believe that the high points of these works (and of others such as *Les nuits d'été* and *Les troyens)* show the composer to be one of the towering giants of music, worthy of being added to The Three B's as a fourth, although at his worst he is perhaps weaker than the other three at theirs. Toscanini called the *Scene d'amour* in *Romeo* the "most beautiful music in the world" and I'm inclined to agree. (Berlioz thought so, too.)

Now, Inbal is no Toscanini; he is not an elemental force like the great Parmesan, but he is a very good Berlioz conductor. He is a precisionist who can bring out a detail you weren't aware of before and his phrasing can be very beautiful. He is also keenly aware of the dynamic gradations of the music and builds the big climaxes very effectively by not making every crescendo a climax. Overall, his conducting has vigor, control, and lyricism in equal proportions. Strangely, he is at his weakest where gossamery lightness and elfin charm are called for, as in the Queen Mab scherzo *of Romeo* and, in *Faust*, the dance of the sylphs and the minuet of the will-o'-the-wisps. Toscanini was pure magic in such scherzando passages, and some of today's conductors do them almost as well. Inbal is at his best in the big, dramatic moments of the *Fantastique,* the *Requiem*, *Roméo,* and the *Te Deum*, where he applies his ability to clarify the score without losing momentum. His soloists and choruses are generally of a high caliber throughout.

The recordings are uniformly excellent and easily equal or surpass the competition; the one exception is the *Requiem,* of which Robert Shaw's recording on Telarc is cleaner and more focused in the "Dies irae" and other complex sections. Elsewhere the Nippon Columbia technique achieves great clarity, realistic textures, and good dynamics.

Recent Classical CD Releases

Composer	Work	Performed by	Label	Code	Date	Producer	Engineer
Bach, J. S.	The "Goldberg" Variations, BWV 988; other keyboard pieces.	Andrew Rangell, piano	***Dorian*** DOR-90138	DDD	10/89	Edwin I. Lawrence	Craig D. Dory et al.
Bartók, Béta also: ***Dohnányi, Ernst von***	Concerto for Orchestra Konzertstück	Seattle Symphony, Gerard Schwarz, cond. János Starker, cello	***Delos*** DE 3095	DDD	12/89 6/90	Adam Stern	John Eargle Andrew Dawson
Beethoven, Ludwig van	Sonata #28 in A Major, Op. 101; Sonata #29 in B-flat Minor, Op. 106 ("Hammerklavier").	Andrew Rangell, piano	***Dorian*** DOR-90143	DDD	1/90	Edwin I. Lawrence	Douglas Brown et al.
	Sonata #30 in E Major, Op. 109; #31 in A-flat Major, Op. 110; #32 in C Minor, Op. 111.	John O'Conor, piano	***Telarc*** CD-80261	DDD	6/90	James Mallinson	Jack Renner
Brahms, Johannes	Sonata #3 in F Minor, Op. 5; Three Intermezzi, Op. 117.	Robert Silverman, piano	***Stereophile*** STPH 003-2	AAD and ADD	1/90	John Atkinson (no kidding!)	Kavi Alexander
	Sonata #3 in F Minor, Op. 5; Sixteen Waltzes, Op. 39; Three Intermezzi, Op. 117.	Antonin Kubalek, piano	***Dorian*** DOR-90141	DDD	1/90	Randall Fostvedt	Douglas Brown et al.
Bruckner, Anton	Symphony #4 in E-flat Major, Haas edition.	Cleveland Orchestra, Christoph von Dohnányi, conductor	***London*** 430099-2	DDD	10/89	Paul Myers	Colin Moorfoot
	Symphony #6 in A Major	Cincinnati Symphony Orchestra, Jesús López-Cobos, conductor	***Telarc*** CD-80264	DDD	2/91	Robert Woods	Jack Renner
Chopin, Frédéric	The Four Scherzi Two Etudes Four Mazurkas	Ivan Moravec, piano	***Dorian*** DOR-90140	DDD	11/89	Randall Fostvedt	Douglas Brown et al.
Diamond, David	Romeo and Juliet (1947) Psalm (1936) Kaddish (1989) Symphony #3 (1950)	Seattle Symphony, Gerard Schwarz, cond. János Starker, cello (in the Kaddish)	***Delos*** DE 3103	DDD	10/90 9/90 1/91 9/90	Adam Stern	John Eargle
Dvorák, Antonín also: ***Janácek, Leos***	Symphony #6 in D Major, Op. 60 Rhapsody for Orchestra "Taras Bulba"	Cleveland Orchestra, Christoph von Dohnányi, conductor	***London*** 430204-2	DDD	3/89	Paul Myers	Colin Moorfoot John Pellowe
Grieg, Edvard	Piano Concerto in A Minor, Op. 16 Lyric Suite, Op. 54 Holberg Suite, Op. 40	Bella Davidovich, piano Seattle Symphony, Gerard Schwarz, cond.	***Delos*** DE 3091	DDD	5/89	Adam Stern	John Eargle

The Music	The Performance	The Recording
Harpsichord masterpiece that should be played on the piano only by an artist who can plead a special case for that alternative, e.g., Glenn Gould.	Rangell is good, but in the Goldberg he is no Gould. The latter's X-raying of the counterpoint, his motor energy, his exhilarating rhythm are missing. Lyricism and lots of pedal instead.	Slightly more reverberant than Craig Dory's best—and generally somewhat earlier—piano recordings in the Troy Savings Bank Music Hall but still outstanding.
The Bartók (1943) is one of the most brilliant, beautifully crafted, and accessible orchestral works of the 20th century; the Dohnányi (1905) is an elegant, not at all modern miniconcerto.	A sleeper. One of the most meticulous and idiomatic performances of the Bartók work ever recorded, lacking only the ultimate orchestral virtuosity. Starker is superb in the Dohnányi.	What John Eargle and his team do in Seattle constitutes some of the finest orchestral recording work in the world, and this is as good an example of it as any. Absolutely beautiful.
The Op. 101 is merely great; the "Hammerklavier" is one of the cornerstones of the piano literature, a gigantic work conceptually and a challenge to the Beethoven interpreter.	Billed as the heir to Schnabel/Serkin, a very classy musician disappoints here with merely fine performances. Richard Goode, with less technique, gets more out of this music.	Superbly clean piano sound with tremendous attack transients and dynamics, but I'd prefer less Troy Savings Bank Music Hall "bloom," especially in the heavily pedaled passages.
Three of Beethoven's most sublime utterances, in the same exalted class as the late quartets. My absolute favorites in the piano literature for listening alone, without any interruptions.	Marvelous playing, in perfect taste; beautiful phrasing without any eccentricities, scrupulously faithful to Beethoven's intentions. Only that final touch of profundity is missing.	Natural, perfectly balanced piano sound with just the right amount of hall ambience (Mechanics Hall, Worcester, Massachusetts). Nothing exaggerated, just as in the performances.
"...labored and bombastic proclamations... stretches of arid manipulation...sentimentality and pretentiousness.. .the result of the small-scale artist's determination to write nobly and and greatly, his attempt to accomplish this by inflating something small into something big and by producing out of technique what does not take shape out of emotional impulse." —B. H. Haggin (1945)*	Very competent but undistinguished playing, with a bar-by-bar, phrase-by-phrase approach, without much feeling for the longer arch of the entire movement being played.	An exercise in analog-with-vacuum-tubes cultism. The hiss is intolerable by today's standards, the dynamic range limited. And the piano is *the* most digital-friendly instrument!
	What a difference! Sensitive and at the same time virtuosic playing, with obvious regard for the long line and the total architectonics. Volume I of Brahms's complete piano music.	Dorian's best piano sound, meaning the best piano sound recorded by anyone anywhere. The unique acoustics of the Troy Savings Bank Music Hall are properly used here.
One of the best Bruckner symphonies and probably the most immediately appealing to the non-Brucknerite. The "Hunting Scherzo" is one of my favorites.	This performance has the Austro-German diction required to give shape to Bruckner's rambling phrases. Magnificent orchestral playing; preferable overall to Chailly, also on London.	Very good Decca/London-type sound with clean brass fortissimi; Colin Moorfoot did a better job here than John Pellowe two months earlier with Dvorak (same hall, same forces).
Few consider this their favorite Bruckner work but it tends to grow on you. It certainly displays all of the composer's strengths *and* weaknesses. Stupendous brass passages.	Meticulous, lucid performance without much tonal refinement or ripsnorting virtuosity; even so, the music makes its statement effectively, and the brasses are duly assertive.	Bruckner needs more acoustical elbowroom than Kunzel/Pops; Cincinnati's Music Hall seems a bit constrictive here, making an otherwise classic Telarc recording slightly raucous.
"Scherzo" means a biggie in the Chopin vocabulary; "mazurka" means a cameo (ethnic subdivision); an "etude" can go either way. All of it superb, unforgettable piano music.	Moravec is a world-class virtuoso, but in a 3-way blind shoot-out with Rubinstein and Van Cliburn in the Op. 39 scherzo, two professional musicians and I placed him a weak 3rd.	The same comments apply as in the case of the Brahms/Kubalek above. Couldn't be cleaner or more dynamic—why aren't all Dorian piano recordings in the TSBMH like this?
The rediscovery of David Diamond continues to be a rewarding experience. Where were the fans when this highly listenable and superbly crafted orchestral music first appeared?	Gerard Schwarz has become the "official" interpreter of David Diamond, recording under the latter's direct supervision, so I can't even imagine any other way of playing this. Great.	Diamond's sonorities are grist for John Eargle's mill; orchestral recordings don't get much better than this. The "hi-fi" scoring spellbinds without any engineering vulgarities.
The Dvorak Sixth deserves to be played more often; it has a stunning scherzo movement (Czech *furiant).* The Janácek music is highly original, colorful, and powerful.	Terrific performances; the Cleveland Orchestra sounds like one of the world's best here, which of course it is, but their playing under Dohnányi is particularly inspired in this music.	Possibly the best orchestral sound I've heard from Decca/London so far—airy, dynamic, not at all overbright. (Telarc, Delos, Denon, Dorian, etc., had better look to their laurels.)
War-horse romantic piano concerto, far from my favorite but can be exciting. The two suites (the Holberg for string orchestra only) are very listenable and rather forgettable.	The concerto is played in a careful, restrained manner, exactly the way it shouldn't be. (Compare with, say, Rubinstein.) The suites are well played without any great distinction.	John Eargle sets his mikes to avoid harshness in the brightest, not the average, passage. In that respect he stays ahead of all competition and then he matches them in all others. Super.

*I don't share this view completely, but it's quite insightful and far from being off the wall. Then again, even Haggin admitted that "the Symphony No. 4 is a magnificent work" and that the Variations on a theme of Haydn are a "superb example" of what Brahms does best.

Recent Classical CD Releases *(continued)*

Composer	Work	Performed by	Label	Code	Date	Producer	Engineer
Handel, George Frideric	Water Music: Suite in F Major; Suite in G Major; Suite in D Major	Orchestra of St. Luke's, Sir Charles Mackerras, conductor	***Telarc*** CD-80279	DDD	3/91	Elaine Martone	Jack Renner
Hanson, Howard	Symphonies #3 and #6 et al.	Seattle Symphony, Gerard Schwarz, cond.	***Delos*** DE 3092	DDD	5/90 10/89	Adam Stern	John Eargle
	Symphony #4, Op. 34 ("Requiem") et al.		***Delos*** DE 3105	DDD	2/91	Adam Stern	John Eargle
Haydn, Joseph	Symphony #100 in G Major "Military" Symphony #103 in E-flat Major "Drum Roll"	Orchestra of St. Luke's, Sir Charles Mackerras, conductor	***Telarc*** CD-80282	DDD	3/91	James Mallinson	Jack Renner
Hoist, Gustav	Suite #1 in E-flat; A Moorside Suite; Suite #2 in F; Hammersmith, Prelude and Scherzo.	Dallas Wind Symphony, Howard Dunn, conductor	***Reference Record-ings*** RR-39CD	DDD	6/90	J. Tamblyn Henderson Jr.	Keith O. Johnson
Janácek, Leos	Glagolitic Mass	Atlanta Symphony Orchestra & Chorus, Robert Shaw, cond., with quartet of soloists	***Telarc*** CD-80287	DDD	11/90	James Mallinson	Jack Renner
also: ***Dvorák, Antonín***	Te Deum, Op. 103						
Liszt, Franz	Variationen über Bach; Liebesträume; Bénédiction de Dieu dans la solitude; Funérailles.	Michel Dalberto, piano	***Denon*** 81757 9289 2	DDD	6/90	Yoshiharu Kawaguchi	Hiroshi Goto
Mahler, Gustav	Symphony #8 in E-flat Major ("Symphony of a Thousand")	Atlanta Symphony Orchestra & Chorus, Robert Shaw, cond., with soloists & extras	***Telarc*** CD-80267	DDD	4/91	Robert Woods	Jack Renner
Mozart, Wolfgang Amadeus	Symphonies #26 in E-flat Major, K. 184; #29 in A Major, K. 201; #39 in E-flat Major, K. 543.	Sinfonia Varsovia, Emmanuel Krivine, conductor	***Denon*** 81757 9202 2	DDD	7/90	Yoshiharu Kawaguchi	Hiroshi Goto
	Requiem in D Minor, K. 626; Eine Kleine Freimaurer-Kantate, K. 623.	Boston Early Music Festival Orchestra & Chorus, Andrew Parrott, cond., with soloists	***Denon Aliare*** 81757 9152 2	DDD	6/90	Elizabeth Ostrow	Henk Kooistra
Mussorgsky, Modest	Pictures at an Exhibition (orch. Ravel) Night on Bald Mountain (orch. Rimsky-K.)	Atlanta Symphony Orchestra, Yoel Levi, conductor	***Telarc*** CD-80296	DDD	1/91	Robert Woods	Michael Bishop
Ravel, Maurice	Daphnis and Chloë (complete ballet)	Seattle Symphony and Chorale, Gerard Schwarz, conductor	***Delos*** DE 3110	DDD	9/90	Adam Stern	John Eargle
also: ***Diamond, David***	Elegy in Memory of Maurice Ravel				1/91		
Strauss, Richard	Don Juan, Op. 20	Vienna Philharmonic Orchestra, Andre Previn, conductor	***Telarc*** CD-80262	DDD	11/90 12/90	James Mallinson	Jack Renner
	Don Quixote, Op. 35	Rainer Küchl, violin; Heinrich Koll, viola; Franz Bartolomey, cello					

The Music	The Performance	The Recording
If you've ever listened to baroque music, you know this. If you don't know it, it may be the one piece that will arouse your interest in baroque music. Accessible and irresistible.	Very good, musicianly, traditional type of performance with modern instruments, but without that extra measure of vitality and bounce that can raise this music to incandescence.	Almost anything sounds good in New York's American Academy and Institute of Arts and Letters, as does almost anything ever recorded by Jack Renner. Now put the two together...
Hanson's later symphonies are gnarlier and not quite as immediately appealing as the popular #2, but each has great thematic and harmonic strength plus brilliant orchestration.	Schwarz has made this music one of his specialties and he conducts it authoritatively, with fine playing by the Seattle band. Hanson's own recordings (1950s) are the competition.	Seattle/Schwarz/Stern/Eargle on Delos is money in the bank, soundwise. I can't imagine anything better in terms of "texture" or "structure," to use John Eargle's own terminology.
Two of the very best of Haydn's 104 symphonies, the "Military" being particularly favored by audiophiles because of the "Turkish" percussion battery, but the 103rd is even greater.	The same forces in the same hall the same week did an even better job with Haydn than with Handel above. Good orchestral playing and quite deliciously inflected phrasing.	The same hall as in Handel above sounds just a bit too small and unreverberant for a full orchestra, even a Haydn-sized one, but Jack Renner still managed to obtain a lovely sound.
Nobody wrote better band music than Gustav Hoist, and these works show the same imagination and craftsmanship in the use of woodwinds, brasses, and percussion as *The Planets*.	Very nice playing but not quite in the same league as The Cleveland Orchestra's "Symphonic Winds" under Frederick Fennell in the two Suites, back in 1978 (Telarc).	Keith Johnson had the advantage of 12 years and the superb new Dallas hall over the old Telarc recording with Soundstream; he wins, but only by a narrow margin (highs a bit hot).
The Mass according to Janácek is a savage, immensely colorful affair, as powerful as the *Sinfonietta* but—to me—not very devotional. The Dvorak is more conventionally beautiful.	Shaw is incapable of doing this sort of thing less than superbly; it's great, but native Slavs must be rolling on the floor hearing performers with Anglo names singing in Old Slavonic.	Jack Renner recording choral works conducted by Robert Shaw in Atlanta is the very definition of state-of-the-art in that category, and this CD proves it once again. Perfect.
liszt is a much greater composer than some antiromantics give him credit for. The Bach variations, for example, written when he was in his early fifties, are absolutely magnificent.	Dalberto is French, 36 years old, and has a huge technique. He performs these pieces with great style, musical sensitivity, and lucidity; one couldn't ask for more. World class.	Leans toward the "he is here" piano sound as against "you are there." Minimal ambience but gorgeous Steinway tone and wide dynamic range. Denon still pre-emphasizes all CDs.
Not my favorite Mahler; despite the gigantic concept and all-out attempt at heaven storming, it sounds contrived to me and ultimately unconvincing. Some beautiful passages, yes.	Three levels of performance here: beautiful work by the choral forces, good but not great singing by the soloists, and competent but rather routine conducting by Shaw.	A CD landmark: 80 minutes of music on a single disc (79:39 to be precise). Fantastic! And the sound (monitored on the new Waveform speakers) is first-chop Renner/Atlanta.
The best of the youthful Mozart (K. 201) and the best of the mature Mozart (K. 543)—what more could one ask for? This program goes from the exquisite to the sublime.	Krivine is a very gifted conductor, and the Warsaw orchestra is an excellent one, although perhaps not quite as responsive as the Philharmonia he used earlier in this Mozart series.	The Denon technique of recording an orchestra (see Issue No. 12) seems to work equally well in Frankfurt and in Warsaw; the sound here is beautifully smooth, clean, and focused.
Mozart's legendary unfinished last work, magnificent up to the point where he left off; then it goes clunk where his pupil Süßmayr took over from scratch (in the last four sections).	One of those problematic performances: very nice, very musical, but not good enough to compare seriously to the best in a crowded field. Parrott is undoubtedly a fine musician.	Live recording in a Boston church, not by the usual Denon team. As such it's very successful, quite comparable to the standard Denon product, which is right up there with the best.
War-horses, but what war-horses! Is there a conductor—good, bad, or indifferent—who didn't want to show the world what he could do with these showpieces?	Very fine playing by the Atlanta orchestra, with very precise control by a conductor who seems to be an old-school podium disciplinarian. In this music, it works very effectively.	Is Michael Bishop is trying to show the boss, Jack Renner, how a classic Telarc recording with just Schoeps MK.-2L mikes ought to sound? Demo quality—an audiophile must!
This is quintessential Ravel; if you like him, you must own at least one recording of the complete ballet (not just the suites). I admire the craftsmanship more than I love the music.	Schwarz takes this very seriously and makes an all-out effort to turn in the performance of a lifetime. The results are very fine indeed; let's say he is at least an honorary Frenchman.	John Eargle offers the alternative to the hard-edged, incisive Telarc school: softer, but just as transparent, more panoramic, maybe even more real. If you press me, he is my favorite.
I'm a sucker for the worst of Strauss, so what can I say about his masterpieces? The two Dons represent his youthful and lifetime best, respectively; the repertory is unimaginable without them. War-horses, yes—but I'll trade you a Mahler symphony for each.	No orchestra in the world plays this sort of music more beautifully than the Vienna Philharmonic—absolutely breathtaking. Previn's conducting, on the other hand, is lethargic and uninflected as compared to Toscanini's or Reiner's, although sure-handed and idiomatic.	The sound, as recorded in the Musikvereinsaal in Vienna exclusively with Sennheiser microphones, is so gorgeous that I almost prefer to listen to Previn's good-but-not-great performances than any of the others. Inner details emerge as in no rival recording.

Recent Classical CD Releases *(continued)*

Composer	Work	Performed by	Label	Code	Date	Producer	Engineer
Strauss, Richard also: ***Françaix, Satie, Ibert***	Oboe Concerto (Miscellaneous works with oboe, reissued)	John de Lancie, oboe Chamber Orchestra, Max Wilcox, conductor (Previn/London, reissd.)	***RCA Victor Gold Seal*** 7989-2	DDD ADD	5/87 8-9/66	Max Wilcox (P. Dellheim)	Timothy Martyn (K. E. Wilkinson et al.)
Stravinsky, Igor	The Firebird (Suite, 1919 version) Petrouchka (1947 version)	Baltimore Symphony Orchestra, David Zinman, cond.	***Telarc*** CD-80270	DDD	3/91	Robert Woods	Michael Bishop
	Le Sacre du Printemps (The Rite of Spring) Symphony in Three Movements	New York Philharmonic, Zubin Mehta, conductor	***Teldec*** 2292-46420-2	DDD	9/90	Max Wilcox	Max Wilcox
	Le Sacre du Printemps Symphony in Three Movements Oedipus Rex (et al.)	The Orchestra of St. Luke's, Robert Craft, conductor, with soloists (Vol. I of a new series)	***Music Masters*** 01612-67078-2	DDD	1991	Gregory K. Squires	Gregory K. Squires
also: ***Prokofiev, Sergey***	Le Sacre du Printemps (The Rite of Spring) Scythian Suite, Op. 20	Dallas Symphony Orchestra, Eduardo Mata, conductor	***Dorian*** DOR-90156	DDD	2/91	Douglas Brown	Craig D. Dory
Suk, Josef	Asrael: Symphony in C Minor, Op. 27	Royal Liverpool Philharmonic Orchestra, Libor Pesek, conductor	***Virgin Classics*** VC 7 91221-2	DDD	5/90 and 12/90	John H. West	Mike Clements
Tchaikovsky, Peter Ilyich also: ***Arensky, Anton***	Piano Trio in A Minor, Op. 50 Piano Trio in D Minor, Op. 32	The Rembrandt Trio Valery Tryon, piano G. Kantarjian, violin C. Bloemendal, cello	***Dorian*** DOR-90146	DDD	5/90	Antonin Kubalek	Craig D. Dory
Weber, Carl Maria von also: ***Brahms, Johannes***	Clarinet Quintet in B-flat Major, Op. 34 Clarinet Quintet in B Minor, Op. 115	Eddie Daniels, clarinet The Composers String Quartet	***Reference Recordings*** RR-40CD	DDD	9/90	J. Tamblyn Henderson Jr.	Keith O. Johnson

Recent Pop and Jazz CD Releases

Composer	Title	Artist(s)	Label	Code	Date	Producer	Engineer
Mays, Bill also: ***Gershwin, Mingus, Monk, Rodgers et al.***	One to One 2	Bill Mays, piano Ray Drummond, acoustic bass	***dmp*** CD-482	DD	12/90	Bill Mays Tom Jung	Tom Jung
Mintzer, Bob also: ***Gershwin, Miller, Youmans***	Art of the Big Band	Bob Mintzer, Randy Brecker, Peter Erskine, Chuck Loebet al.	***dmp*** CD-479	DD	9/90	Bob Mintzer Tom Jung	Tom Jung
Wilson, Meredith	The Music Man	Timothy Noble Kathleen Brett Doc Severinsen Cincinnati Pops/Kunzel	***Telarc*** CD-80276	DDD	4/91	Robert Woods Elaine Martone	Jack Renner
(Various)	The Forward Look	Red Norvo Quintet	***Reference*** RR-8CD	ADD	12/57	Keith O. Johnson	Keith O. Johnson

The Music	The Performance	The Recording
Lovely work in four movements for oboe and small orchestra by the 81-year old Strauss, dedicated to "an American soldier," who was John de Lancie himself, pilgrimaging in 1945.	Max Wilcox steps successfully from the console to the podium to conduct an incisive, flowing, carefully controlled performance. De Lancie isn't getting younger but is still great.	This is a four-year old recording, finally issued in 1991, combined with 25-year old analog cuts. It deserved better. The sound is typically Max Wilcox: clean, lucid, unexaggerated.
The Firebird suite contains all the good stuff from the ballet, Stravinsky's first great leap past Rimsky/Debussy. The 1947 pared-down version of Petrouchka is too sparse and dry.	The Baltimore orchestra plays with amazing instrumental refinement and precision; Zinman has done wonders here. The Firebird performance, especially, is as good as you'll find.	Once again, Michael Bishop is trying to one-up the boss and succeeds; this is definitely demo-quality material. The Infernal Dance in the Firebird suite is one of my current demos.
I'm giving myself elbowroom here to sound off on one of my favorite subjects. If I were pressed to award some kind of Olympic gold medal to just one 20th-century work of music, it would have to be the Sacre, without a doubt (*pace* Bartók and all the others). I find it absolutely uncanny that this music was born in the mind of a 1911 composer. Where did these cadences, these sonorities, this musical language come from? Truly sprung from the head of Zeus, like Pallas Athene. In a modern performance, the miracle of the music's originality and savage power must not be allowed to settle down to a cozy Debussyan idiom.	The problem in a performance of the Sacre is to project savagery and unbridled passion within a framework of refined orchestral sound and with great precision in the virtuoso passages. Robert Craft, who as Stravinsky's confidant obviously knows the idiom, is by far the most powerful and convincing here in terms of rhythm, phrasing, and salient detail, but his orchestra is a bit rough. Mehta is strangely uninvolved, not nearly as authoritative as just a few months earlier in Mahler, Hoist, and Sibelius. Mata tries hard to deliver a definitive performance with his excellent orchestra, but he is a little too neat and deliberate.	I always knew that Craig Dory's first orchestral recording would be super special—and is it ever! Aided by the wonderful acoustics of the Meyerson hall in Dallas, the sound is so rich, powerful, lucid, and three-dimensional that even Delos and Tel arc must be worried. Max Wilcox, in the New York recording, is fighting the acoustics of the Manhattan Center, where Stravinsky's orchestration doesn't seem to fit too happily. Even so, he ends up with very creditable sound. The Robert Craft recording is very dry but extremely clear and detailed, just like his conducting. I suppose that's what Igor would have liked.
A big, sprawling, late-romantic, 62-minute symphony by one of the second-team Czechs (still a damn good team). Asrael is the Muslim angel of death—you get the 1905 flavor.	The Liverpool orchestra could also be characterized as a good second team; only the ultimate refinement is missing. Pesek takes this music very seriously and makes the most of it.	Interesting technique—very high definition, with every detail leaping out at you, etched but undistorted. Some will call it state-of-the-art; others will find it just a wee bit irritating.
The romantic chamber repertory is exemplified here with the best work of a great and a minor composer, respectively, in the trio format. Who could possibly not like this lovely music?	Canadian musicians named after a Dutch painter? Still, they're very good. I prefer, however, the more fiery, romantic playing of Cardenes/Golabek/Solow on Delos in this music.	The acoustics of the Troy Savings Bank Music Hall are again given too much billing here, but not enough to spoil a basically lovely sound. The Delos version is more focused sonically.
Behind the Mozart clarinet quintet, these two are probably the best candidates for the silver and bronze. The Weber is upbeat; the Brahms is a late work, autumnal in flavor. I love both.	Daniels is a top-notch clarinetist (classical and jazz), and the Composers quartet is first-rate, but the ugly recording makes it difficult to gauge the beauty of these alert performances.	I can't believe this! Intolerably bright, edgy, close-up sound—and in the Troy hall, of all places. Maybe RR cares only about the LP version and is precompensating for analog losses.

The Music	**The Performance**	**The Recording**
Elaborate and very cool jazz improvisations on just two instruments, piano and bass, about as lean and pure as you can get, and often very beautiful. Sequel to *One to One* (also on *dmp).*	Two outstanding musicians doing their best. Bill Mays is a very imaginative jazz pianist and Ray Drummond is simply awesome on bass. Real smoke-filled cellar-club stuff.	I wish all classical recordings of the piano and one stringed instrument were as crystal-clear and natural as this jazz recording by Tom Jung. They don't come any better.
The big-band sound brought up to date (or at least up to the 1960s), with original stuff by Bob Mintzer interspersed with great oldies.	Very good playing by highly professional musicians, better than what was the norm in the big-band era, but I'll take Glenn Miller anyway.	In terms of instrumental presence, one of the most amazing recordings known to me. Tom Jung is second to none in this sort of thing.
Hit musical from the 1950s, cornball and mock-sophisticated by turns, with lots of brassy band numbers and a few nice songs. "Seventy-Six Trombones" is the well-known theme music.	Expert oompahing by the Cincinnati Pops; very energetic but not particularly charming singing by the principals. I'm sure that Robert Preston was more appealing in the original.	Another of the recent Jack Renner recordings monitored on the Waveform speaker. Spectacular sound, lots of presence, wide dynamic range, but a bit too bright and aggressive.
Twelve classic cuts by the great vibraphonist and his sidemen. Superb jazz, all of it.	Recorded live, without balancing or editing, this is in the highest 1950s jazz style. Great!	One of the earliest stereo recordings, 34 years old and amazingly 3-D and distortion-free.

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