The Audio Critic®

Issue No. 25

Display until arrival of
Issue No. 26.

Look, Ma, no amplifiers! (Well, no amplifiers to buy, anyway. See the loudspeaker reviews.)

Retail price: U.S. $7.50, Can. $8.50

In this issue:

We review a wide assortment of loudspeaker systems, including minimonitors, various full-range speakers, and powered subwoofers from extra small to large.

We continue our notorious survey of the White Hats (good guys) and Black Hats (bad guys) of audio.

Power amps, integrated amps, multichannel receivers, digital processors, CD and DVD players, FM gear, and a minimalist preamp are reviewed in depth.

The end of innocence in audiophilic matters is the aim of a down and dirty orientational think piece.

Plus our usual features, columns (we have added a new one), letters to the Editor, and more CD capsule reviews.

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The Audio Critic is an advisory service and technical review for consumers of sophisticated audio equipment. Any conclusion, rating, recommendation, criticism, or caveat published by ***The Audio Critic*** represents the personal findings and judgments of the Editor and the Staff, based only on the equipment available to their scrutiny and on their knowledge of the subject, and is therefore not offered to the reader as an infallible truth nor as an irreversible opinion applying to all extant and forthcoming samples of a particular product. Address all editorial correspondence to The Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951-0978.

From the Editor/Publisher:

Our efforts to become part of a larger organization (see this same space in the last issue) have so far elicited some interest in the publishing world without producing an acceptable partnership. Thus your graying Editor remains a one-man band—in the laboratory, at the computer, in the writing/editing/publishing loop—putting out a magazine single-handed, from ground zero to camera-ready mechanicals for the printer. If 1 stopped having a life and devoted every waking minute to the job, I could probably cut in half the interval between issues. That interval, as this issue proves, tends to be so long that even half of it is unacceptable, leading to the inescapable conclusion that getting plugged into a larger publishing operation is the only sensible plan. Rest assured, it will happen, maybe soon, maybe later. Meanwhile you hold in your hand two issues' worth—counting the number of test reports, other reviews, and features—for the price of one.

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You may start your subscription with any issue, although we feel that new subscribers should have a few back issues to gain a better understanding of what ***The Audio Critic*** is all about. We still have Issues No. 11, 13, 14, and 16 through 24 in stock. Issues earlier than No. 11 are now out of print, as are No. 12 and No. 15. Please specify which issues you want (at $24 per four).

One more thing. We don't sell single issues by mail. You'll find those at somewhat higher cost at selected newsdealers, bookstores, and audio stores.

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Letters to the Editor

Your letter has a good chance to get published here if it is a relevant commentary on audio issues or the contents of our journal. Please send only typewritten or word-processed text; our scanner software does not recognize hen scratches. Letters 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 18951-0978.

We lead off with an object lesson about the perception of consumer audio by the general (i.e., nontechnical) press. The following exchange of letters between your Editor and Forbes *magazine took place a few months after the publication of Issue No. 24. The letters are self-explanatory.*

Mr. James W. Michaels, Editor
Forbes
New York, NY

Re: "Going vinyl" by Stewart Pinkerton, under Living, *Forbes,* July 7, 1997.

Dear Mr. Michaels:

As a well-wisher of *Forbes,* I feel compelled to point out to you the utter lack of science, professionalism, and accountability in the above article. It seems you have been snookered by Mr. Pinkerton and the agenda-driven tweako/ weirdo audio cultists he hangs out with. He obviously does not know how to network with authorities possessing serious technical credentials.

"...Records sound better than CDs because they're direct analog copies of a musical event, rather than digitized reconstructions." "…A high-quality photograph looks much better than a digital image. The detail is sharper, the color richer." This is a scientific illiterate talking. The Nyquist-Shannon sampling theorem and the principles of quantization provide unequivocal assurance that with 44.1 kHz sampling and 65,536-step quantization as used in CDs no "detail" in the audio spectrum and the audio dynamic range is lost—not an iota, not a scintilla. By any criterion of audio fidelity—frequency response, dynamic range, distortion, signal-to-noise ratio, channel separation, etc.—the CD medium is devastatingly superior to vinyl. Today's LP microindustry is based on nostalgia and the utilization of leftover production facilities, not technical advantages. Mr. Pinkerton's so-called comparative test was a typical salesman-coached exercise in undisciplined subjective expertizing, without any scientific controls (such as level matching, double-blind conditions, etc.).

What I see as the root cause of your editorial problem here is your Rolodex. You and your writers do not appear to have the names and telephone numbers of genuine audio experts. To use a simple analogy, instead of asking the Mayo Clinic about medical questions you have been asking your local health food store. For solid information about audio, you need to be in touch with the Audio Engineering Society (AES), with the Institute of Electrical and Electronics Engineers (IEEE), with tenured professors of electrical engineering, acoustics, physics, and applied mathematics at leading universities, with the top engineering executives of major manufacturers in the field, and so forth. Not with retail stores trying to sell you the stuff, for heaven's sake, or with untutored hobbyists rationalizing their purchases. You cannot imagine the utter contempt and jeering ridicule elicited by Mr. Pinkerton's kind of unscientific journalism from the top professionals and academics I am talking about.

If you are serious about upgrading *Forbes's* capabilities in this area, I shall be glad to help you assemble a better Rolodex. I have all the best names and numbers, and want no compensation other than improved hi-fi reporting in *Forbes.* Meanwhile I am enclosing a couple of clippings from my publication that have some relevance to this matter.

Sincerely,
Peter Aczel
Editor and Publisher
Life Member of the AES

Dear Mr. Aczel:

The typical *Forbes* reader doesn't have a doctorate in acoustics or electrical engineering. But, many of them do spend a good deal of money on hobbies such as audio. Thus we're interested in giving them an insight into the available choices in the market. There's just as much hype coming from the CD enthusiasts as there is from the vinyl advocates. I'll pass your comments along to Mr. Pinkerton.

Thanks for taking the time to write.

Sincerely,
Jim Michaels
Editor
Forbes

I couldn't believe my eyes when I first read the above reply. Apparently, correct information is strictly for Ph.D.s, not for "the typical Forbes *reader." But wait! It turns out that they did listen to me, even if they won't admit it directly. In their December 28, 1998 issue there is a short article by Robert La Franco, titled "Selling sizzle with sizzle." It is about Noel Lee and the Monster Cable success story. The focus is on the business aspects of the audio cable industry, but La Franco also points out that high-priced stereo cable "is a product where most of the value is in the mind of the buyer" and that heavy-gauge lamp cord "affords nearly as much fidelity." Isn't that amazing? It's definitely a first in a business magazine. Am I being presumptuous to take a little credit for* Forbes*'s turnabout?*
—Ed.

The Audio Critic:

I was most pleased to see that you put me on the side of the White Hats. I always tried to be so. I also appreciate your kind words about me in Issue No. 24.

Your list of important things in audio on page 9 [of No. 24] is right on the mark. I have supported these priorities for many years. You state them boldly and succinctly.

After many years I have retired from professorship and from writing or even commenting on audio matters. I still do some consulting in acoustics and digital signal processing. But, my main pastimes are now gardening and astronomy. They are both slow, gracious and relaxing activities most suited to retirement.

While I no longer consider the "tweakos" worth arguing with or worth trying to educate, I wish you well in your continuing battle on the side of truth, right and justice.

Sincerely,
(signed) Dick
R. A. Greiner
Emeritus Professor of Electrical and Computer Engineering

Dick Greiner is one of the towering authorities whose name should be on Forbes*'s Rolodex (see above). They would find him kind, highly approachable, and prompt to respond, but of course they are unlikely to consult him. That the editorial stance of this journal has his complete approval is one of the things that make me sleep better at night than Larry Archibald and John Atkinson (if indeed there is any "truth, right and justice" in this world). Thank you, Dick, for having been a beacon all these years in the murky sea of audio.*

—Ed.

The Audio Critic:

Please permit me a little 3-way linkage. I would like to respond to Dan Sweeney's excellent article on multichannel sound, focusing on speaker directivity and "Can One System Do Both?" etc., in Issue No. 24, and combine those thoughts with some generated from Floyd E. Toole's also excellent article in *Audio* magazine (May & June 1997), especially since Floyd Toole was mentioned in Sweeney's article. It is nigh impossible to treat this subject matter in one letter. At the same time, I hope to shed some light on your query in the review of the Waveform Mach 17, about the missing connection to "the variability of response in the vertical plane" (Issue No. 24, p. 34).

For the most part, speaker designers assume that the listener is seated. This means that the listener will obtain the majority of the combined acoustic output in the horizontal hemisphere from a neutral room. Even when the dispersion model chosen is the point source, stacking bass, midrange and treble drivers vertically on a baffle will result in fairly severe broadband nonlinearity generated in the vertical plane at ever increasing wide angles. These are taken fully into account when the total radiated sound power is compiled. One will then also be able to deduce the directivity index, which now appears to be the current darling measurement for the video crowd.

The listening window, which is a computer-generated curve at the NRC, contains the 15° up and down measurements, along with the 0°, 15° L & R. Measuring up and down this amount will help account for what a standing person will experience at a reasonable distance of about 10 feet from a stereo pair. If a listener insists on standing on a ladder or lying on the floor at this same distance, the audible response from the same product may not be quite as memorable. Waveform does include this measurement, along with 17 separate others, in our owner's manual. What is noteworthy here is that the shape and level of the window matches very closely the on-axis curve, something we have never witnessed before.

With respect to multichannel sound, what needs emphasizing is that nowhere in either article, by either author, is there a persuasive defense enunciated for speakers having differing directivity characteristics for film sound and for the sound of music. Sound is sound. To quote Toole: "In principle, there should be no reason to differentiate between them. Good design is good design." *(Audio,* May 1997, p. 137.) It is not my intention to enter into an audio vs. video debate; the marketplace is already doing that, and to audio's detriment. I also have no theoretical quarrel with a speaker that has constant directivity. It is a desirable attribute, but only after the main horizontal curves have been made flat. The sacrifices made are too great in attaining it, although it now may be possible to have both within the same product. Where is the diffraction model in a square-edged cabinet with a horn driver? Where is the diffraction model in a dual-element, dual-use speaker? Where is the acoustical interference model for dispersion in designs with multiple tweeters or midranges, for that matter? And where are the comparative listening tests that show that a listener would prefer this compromise to any other? Constant directivity is achieved, in my view, at the expense of flat extended frequency response in the three-dimensional sound field, whether one employs horns or multiple drive elements.

Speakers that are designed with a smooth and extended dispersion plot are not done so with the intention to provide ambience by bouncing off the room boundaries, as Daniel Sweeney suggests (Issue No. 24, p.71). That this does occur is usually a direct result of a too-live listening room. Controlled dispersion is executed to emulate the point source concept as defined by frequency, not phase. It is up to the user to provide a suitable room acoustic with a predominantly direct sound characteristic, which will happen as a result of low reverberant decay (0.3 s to 0.4 s) as specified (or buried?) in the literature, but not emphasized nearly enough by those who presented and argued for it in the first place.

Since home theater systems are only in about 10% of North American homes, interesting things begin to appear when the demographics of these owners are examined more closely. The early adopters have more discretionary funds to dispose of. Their homes are typically larger and therefore the rooms are larger too. Not so good, as large rooms inherently have longer reverberant times, which lead to louder echoes. The room may be a showpiece and may have only an area rug on a tiled surface or hardwood floor, as opposed to full broadloom. Not so good either, as the potential for flutter echo between the exposed floor and the ceiling is awaiting a trigger mechanism to explode. The owner may have a ravine lot with no neighbours to the rear, so the need to have privacy drapes on expansive glassed walls is reduced or eliminated. More reflective surfaces (and big ones too), not so good again, almost redundantly so. The main reason for controlling the vertical dispersion in THX speakers, as well as with some of the new constant-directivity models, is that owners of home theatre systems and installers have not come to grips with the needed room acoustic, which would require lowering the room sound and raising the speaker sound. Not easy!

There is also an absolute dearth of discussion regarding the correct use of the centre channel. Virtually all on-screen dialogue in films is summed to this speaker, so that actors sound as if they were piggybacked on top of each other in the same wretched fashion that multitrack music recordings stack vocalists and performers vertically in the centre between the L and R channels. The centre channel is not a universal cure-all for listeners sitting off the central axis. Most importantly in music, but no less so in good dramatic film dialogue, there will have to be more care used in creating phantom images between the L & C and the R & C. Here again, level is more important than phase.

One last point about having five identical channels. If, as Floyd Toole suggests, the side channels shouldn't be exactly like the front channels in timbre because of the outer shape of the ear, then it should be up to the encoding within the system software setup to re-equalize the timbral balance of those side channels for where they are positioned on the wall, by height and by distance, either fore or aft of a centrally located listener with head forward.

The really big picture is the one that contains all of the product measurements, all of the room measurements, and all of the psychoacoustic optimizations. There is a level of sophistication needed here that has never been attempted before. Those who do it first and do it correctly will reap the rewards. Then the only difference that remains will be the recording as art form. Is there the will?

Sincerely,
John Ötvös
President
Waveform

The trouble with all theoretical speaker-design priorities, even if intelligently and knowledgeably argued as in your letter, is that "the proof of the pudding is in the eating," i.e., you have to point to an already existing design that incorporates those priorities before we can all be thoroughly convinced. Loudspeakers are not like amplifiers. In speakers there is no such thing as a 100% hardware implementation of a paper design. "Between the potency/And the existence/...Falls the Shadow" (T. S. Eliot). Sometimes a conceptually flawed design sounds better than a brilliant one as a result of better execution.

Since the Waveform Mach 17 is, overall, the most successful loudspeaker design known to me, I take your priorities very seriously and await eagerly the grand synthesis they wishfully and wistfully point to.

—Ed.

The Audio Critic:

I think that it would show that you had *manners* befitting a serious journal if you didn't use such strong language to describe those lyin', cheatin', dirty, rotten sons o' bitches/bastards!

David Franklin
Edgartown, MA

Who uses strong language ? Not this freakin 'journal! (But then all criticism is perceived as strong language by the criticized. "Don't speak of rope in a hanged man's house," says an old Hungarian proverb, but that doesn't mean rope is a dirty word.)

—Ed.

The Audio Critic:

...I want to reassert forcefully: thank you for *The Audio Critic.* It has brought the clear light of day into an area of consumer life that I had been bumping around in clumsily for all too long. I do not think integrity and clearly developed reasoning and crisp, straightforward prose are anywhere in our culture ready to hand; it is precisely these virtues I find on every page of your Issue No. 24, which is the first issue of your journal I've met; and I cannot be quick enough in subscribing and including amounts for ...back issues...Nos. 16-23.

...Believe me, I will not complain about irregular publication schedules. I tell my students the same warning about when to expect back papers they hand in: if I am to do a good job commenting, they must wait until I am ready. Besides, the rich substance of what I read in this single issue provides me with plenty to reflect on.

As our literature studies try to include more cultural contexts, I find myself always making students alert to the hype of consumerism; and your journal performs that great service in an area where I hear almost no one else speaking.

Thank you again very much.

Sincerely,
Roger Kaye
Department of English
California State University
Chico, CA

Isn't it remarkable that university people endorse and support us so much more consistently than the untutored audio-salon dweebs and magazine-rack moochers? Is there a sociocultural conclusion to be drawn here?

—Ed.

The Audio Critic:

You sure have put some of the fun and sanity back into audio. Your "Good Guys/Bad Guys" article in Issue No. 24 was great, and long overdue. However, one question begs to be asked: Why is it so easy for you to forgive Bob Carver's advertisements? When Bob's ad declares that his Sunfire amp has the soul of a 9-watt triode, he may simply be trying to double his revenue, but he simultaneously glorifies and legitimizes the whole high-end mystical megabucks rip-off routine that your publication has so often denounced. Since most of us longtime audio enthusiasts respect Carver's technical achievements, his ads serve only to confuse us. If Bob is intentionally misleading the audio public in order to increase his income, doesn't that qualify him for "bad guy" status right up there with Dennis Had or Conrad and Johnson?

Terence R. Simmons
New Hope, PA

Bob Carver is a "Good Guy in a White Hat" because he gives you a well-engineered—sometimes brilliantly engineered—product at a fair price. Dennis Had and Conrad-Johnson are "Bad Guys in Black Hats" because they give you dumb-ass tweako engineering at an unconscionably inflated price.

That does not mean Bob Carver's advertising is as good and credible as his Sunfire product. As a former advertising professional I have told him more than once that he should not write his own ads, but he just can't live with anyone else's words—or taste. I happen to know that he secretly congratulates himself for being a hard-nosed marketeer who understands that consumers and dealers must be told what they want to hear, not necessarily the facts. In other words, he is willing to let you buy the right product (his) for the wrong reasons. Not because he is money-hungry—his entire biography would be totally different if he were—but because he thinks that's the way smart marketeers operate. Sometimes he even lapses into borderline tweakspeak just to gain the confidence of the tweako element, although I know for a fact he shares none of their beliefs.

A famous dictum of Bill Bernbach, one of the smartest admen of all time, is that good advertising makes a bad product fail faster. The logic of that is unassailable, but I certainly do not believe the reverse is true—that Bob's bad advertising is making his good product succeed faster—although it probably makes little difference one way or the other. At this point he is a legend, and legends are self-perpetuating. He could quite possibly stop advertising altogether and hardly feel the effect. The bottom line from the Good Guy/Bad Guy perspective is that it would be good for audio if Dennis Had and Conrad-Johnson went away but bad for audio, very bad, if Bob Carver went away. We need him, warts and all.

—Ed.

What follows is a conflation of three letters over an eight-month period from Howard W. Ferstler, author of consumer-oriented books on audio and contributor to The Sensible Sound. *Howard often reminds me of Gertrude Stein's famous quip about Ezra Pound: "a village explainer, excellent if you are a village, but if you are not, not." That was of course patently unfair to Ezra Pound, who was a great poet albeit a nut case, and is also somewhat unfair to Howard, who always fights on the side of scientific rationality in this delirious world of high-end audio. Still, he does tend to be a bit self-important and condescending (which may be just a professional mannerism, as he comes from the university world), whereas I would much rather not be treated like a village (pace Hillary). I have therefore inserted a few brief interruptions into his text, in addition to the longer reply at the end.*

—Ed.

The Audio Critic:

.. .You probably get compliments all the time, but I still thought I'd tell you what a great item Issue No. 24 was. Everything was terrific: Nousaine's contribution, Sweeney's contribution, Rich's contributions—even your [viz., the Editor's] contributions. You do a hell of a job with a three- or four-man writing staff. *[It's really a one-man writing staff with occasional outside contributors who need to be edited.—Ed.]* Also glad to see you tested some cheaper gear and pretty much substantiated what I have said about it—some of it, at least—in two different books on audio hardware: the stuff can work pretty darn good, and impoverished audiophiles can at least have decent sound without feeling ashamed of their electronic hardware or having to hock the silver.

One quibble. On page 25, you state that "a tall planar or line-source speaker puts the listener in the farfield even when he sits fairly close." Well, Peter, that is just not the case. With tall line sources, it is nearly impossible to get into the farfield at nearly any frequency when situated in nearly any home listening room—not at bass frequencies, anyway. The behavior of line sources has been studied to death, but I suggest you read Stanley Lipshitz's paper, "The Acoustic Radiation of Line Sources of Finite Length," presented at the 81st Convention of the AES, November 1986, and hopefully still available as Preprint 2417 (D-4). Stan pretty much came down on the whole concept of tall line sources, mainly because when sitting close to them the top and bottom parts of the line will be radically out of phase with the center at a multitude of frequencies. Roy Allison also dealt with the subject in the BAS *Speaker* (August/September 1984), and Roy and Stan did a give and take on the subject in the December 1984/January 1985 and February/March 1985 issues. Short, controlled-length lines (as with some THX speakers) can work OK, but really long lines can cause problems unless they are placed very carefully and the listener sits just so.

Another quibble. In your review of the Sunfire subwoofer (see page 33), you note that "the Sunfire has the inherent ability to take advantage of the quasi-horn effect of the corner; the bigger subwoofers much less so." Well, it may seem that way, but at the distances involved (a typical 18-inch sub might have its driver centered a foot from two boundaries and two feet from the third) and the frequencies involved (usually below 100 Hz) the drivers of the Sunfire and the bigger unit are both acoustically close to those reflecting surfaces. Indeed, at *very* low frequencies corner reinforcement is probably no better than midwall reinforcement in normal-sized rooms, as long as you do not take the inherent structural stiffness advantage of the corner into account.

Finally, no doubt like a lot of your readers, I am still not sure of what a "wave launch" is and how it impacts the subjective performance of decent, wide-dispersion speaker systems operating in normally reverberant listening rooms—with those speakers placed anywhere other than close to and aimed directly at the listener.

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...On page 71 [of Issue No. 24], in his fine essay on multichannel music formats, Dan Sweeney notes that the parameters for THX speaker performance mandate that "wide-dispersion designs for the front speakers are to be positively avoided in a multichannel playback system." He goes on to note that "THX speakers sound identifiably different than the quasi-point-source radiators that comprise the bulk of the well-regarded audiophile music speakers." Finally, on page 72, he pretty much comes out and says that video-oriented speakers cannot match the integrity of speakers designed for two-channel audio use, and that the build quality of video speakers often does not match that of audiophile systems.

Well, I should point out that in terms of dispersion the only major difference between THX-certified speakers and many audiophile models involves the vertical angle. As far as horizontal dispersion is concerned, typical THX models are a match for many so-called audiophile models, particularly those that employ rather large drivers for the midrange. Dan should have noted that limited vertical dispersion will attenuate ceiling and floor reflections that will possibly muddy the clarity of both video and audio-only playback material. There is nothing inherently wrong with limiting vertical dispersion a bit, although, admittedly, the short line-source configuration can cause lobing anomalies that might be a problem if the listener is off the vertical axis somewhat. However, those lobing anomalies are slight, compared to the ones found in typical, high-end, panel-type speakers.

Being "identifiably different" (page 71, again) does not by necessity make THX speakers sound less accurate than so-called point-source audiophile speakers. Indeed, the differences noted may be the result of improved fidelity. In addition, build quality is something that a manufacturer can adjust with any kind of speaker—THX-certified or audiophile. Lots of so-called high-end speakers with very expensive drivers, massively built cabinets, and exotic crossovers, as you most certainly realize, do not sound all that hot.

Perhaps the best compromise in a speaker for both audio-only and home-theater use would be a design like the discontinued Allison IC-20. That system had two vertical midrange-tweeter-tweeter-midrange panels angled 45° on either side of the forward axis, and the result combined a THX-style vertical radiation pattern with an extremely wide horizontal radiation pattern. Of course, mating a center speaker to that stereo pair would require building some kind of customized model—with Allison components—which I am happy to say I have done. *[So yours is the biggest, right?—Ed.]*

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...In assorted editorial commentaries over the years, *The Audio Critic* has noted with varying degrees of dismay and passion how so many of those involved with audio journalism appear more interested in tweakish speculation than in scientific analysis or even in music. Those commentaries have pretty much been on the mark, in terms of what is going on, but not always exactly at dead center, in terms of the motivations involved or the consequences of those motivations. *[Your commentaries, on the other hand, are "at dead center," right?—Ed.]*

It is interesting that audio, which deals with something so esoteric as the subjective perception of music, has divided into two, for the most part polarized, camps. On the one hand, we have "technophiles," who thrive on brass-tacks analyses and see the discipline as simply an offshoot of science and technology in general. Many audio technophiles are as much at home with physics, computer science, or rocket science as audio, and their audio magazines of choice are probably the *JAES* and *JASA*, and, well, *TAC. [Isn't that "well" a little condescending? See what I mean?—Ed.]* Indeed, computer science has probably drained off a number of talented individuals who would otherwise be heavily immersed in audio technology, and because of this defection of talent our hobby is the worse for wear. At the other extreme, we have a coterie of mystics who often appear to encompass a much wider audience, one that thrives on fantasy and demands a certain amount of obscurity in their audio lives, and I imagine their day-to-day lives as well. That the latter group is willing to spend a lot more money on hardware than the former has certainly not gone unnoticed in journalistic circles. Both groups profess an interest in music, although members of both camps are often more interested in hardware than qualities of performance.

I know of no other "high-tech" hobby that has participants of such widely varying temperaments and attitudes. It is truly a peculiar situation.

In any case, you previously seem to have divided the subjectivist writers who appeal to this surprisingly large, mystic-oriented group into two sections: *The Con Men* (those who know the truth but keep it hidden for monetary or power-trip reasons) and *The Dupes* (those who really believe what they say). It is clear that you consider the former category to be the larger one, with only a handful real crazies occupying the duped-journalist category.

However, it is possible that there is more to the world of subjective-audio journalism than what is indicated by this kind of breakdown, and I think that you and a number of other rationalist critics (including many individuals who have written letters to your magazine about the problem) may be missing an important point concerning what a lot of subjectivist writers, and their editors and publishers, are doing (and are aware that they are doing) when they do seat-of-the-pants equipment evaluations and propose arcane theories of high-end hardware performance.

Indeed, I think we can actually go so far as to postulate a third subjective-journalist category, and it is just likely that it is a grouping that encompasses a sizable number of those who write about the subject of high fidelity, including both The Con Men and The Dupes, whether they write for mainstream or fringe journals: *The Entertainers.*

You see, a great many audio-journalistic entertainers are not being paid to impart "knowledge" at all (be it the brass-tacks oriented stuff the technophiles crave, or even subjectivist-oriented fluff), because a surprising number of readers are simply not interested in technical information—even subjectivist, pseudo-scientific, seat-of-the-pants technical information. The fact that so many different high-end products have received differing reviews (with some conclusions being strongly at variance with others), without the subjectivist readership protesting on a large scale, is an indication that a lot of those individuals really are not assimilating much of anything at all.

Let's face it, most subjectivist audio writers are to a great extent employed to intensify the prejudices and insecurities, or massage the egos, of individuals who, really deep down, couldn't care less about the realistic performance of audio hardware. Those readers are interested in poetry, not specifications. They want to know that a given piece of hardware imparts depth, spirit, sweetness, meaning, profundity, truth, and soul to a recording—not that it merely reproduces an electrical input signal or even accurately reproduces a recorded event. For those people, intellectualizing about audio is spiritually corrosive.

Indeed, if such readers cared about the reproduction of sound, they would not be captivated by single-ended tube amplifiers and the LP record, nor would they agonize about wire and cable, coat their CDs with oil, or clamp their components in heavy equipment racks. As a consequence, most fringe-type audio writers are paid to expertly mesmerize their readers, not give them any kind of information—even wrong information. This is why certain magazines can print pages of test data and then make judgments about the equipment that are totally unrelated to what the graphs indicate, why certain columnists can talk about putting air bladders under amplifiers or rocks on top of them to make them sound better, and why those who review CD players and amplifiers for any number of wildly weird audio magazines can wax enthusiastic about pace, rhythm, and musical truth in their written commentaries. Subjectivist writers are paid to do the same thing music does: tickle the emotions.

I believe that without the phenomenon of audio-journalistic entertainment, a sizable percentage of the readership of many audio publications would defect (some fringe journals would no doubt be completely wiped out), and both the software and hardware industries (particularly the high-end community) would suffer. In addition, without speculative journalistic entertainment some audio buffs would probably quit the pastime completely. Without the mythology, the hobby itself might shrink to an even smaller size than it is today, although those left behind might gain the hobby a little intellectual respect.

It is unfortunate that this little diversion called audio, which in the beginning, several decades ago, was dominated by the brass-tacks crowd, has allowed this Frankenstein monster to be created. However, we are stuck with it and rationalist enthusiasts like you and a handful of other engineering-oriented types (and me, too, I suppose) will just have to learn to roll our eyes and live with the way things are.

"Men judge of things according to their mental disposition and rather imagine than understand."—Spinoza, *Ethics.*

Sincerely,
Howard W. Ferstler
Tallahassee, FL

I should have been much more specific and precise about planar and line-source speakers. Imagine a rectangular listening room one of whose shorter walls is an electrostatic sandwich, from to floor to ceiling and wall to wall, driven in phase over its entire surface. You would be in the farfield—mono of course—no matter how close you were to that wall because an unvarying planar wave front would be traveling down the entire length of the room. The same would be true, but in one dimension only (mea culpa), of a floor-to-ceiling line source from which a cylindrical wave front is traveling down the length of the room. Yes, Howard, I know that typical free-standing planar speakers and free-standing line-source speakers suffer from various interference effects. I was just trying to emphasize that they represent quite different theoretical models than point-source speakers.

As far as the corner placement of subwoofers is concerned, the wall distances for the 11-inch Sunfire are about half of those for the big subs; in other words, the Sunfire is twice as deep into the corner linearly and eight times as deep volumetrically. I am not enough of a physicist/acoustician to tell you exactly how that affects the impedance matching to the air load seen by the driver(s), but I don't have to. The drastically reduced distortion (as compared with placements seeing larger solid angles) tells the story. It's hard to argue with the THD curves.

What is wave launch? I use the term to mean the three-dimensional geometry of wave fronts emerging from a loudspeaker prior to reflections. In my fairly well-deadened room wave launch defines a considerable part of a speaker's sonic signature; those who live in glass houses will probably agree with you because the specifics of the signature will be lost in the reverberant acoustic soup.

The controversy regarding audio-only vs. home-theaters speakers is, I think, adequately addressed in John Ötvös's letter above, to which I have nothing to add at this point because I basically agree with it.

Your entertaining analysis of the phenomenon you call The Entertainers elicits no strong disagreement from me, but I don't believe it is "exactly at dead center." You say I consider The Con Men to be a larger category than The Dupes, but that's not so. I believe that the majority of the tweako pundits are True Believers, i.e., Dupes. They may wish to entertain while expressing their true beliefs, but their principal motivation is to defend the Truth, smite the Infidel (you and me), and give proof of their exquisite taste and hearing acuity. Most of them are poorly paid; their chief reward is the opportunity to play with expensive toys. But if you happen to be right and The Entertainers know that it's all just B.S., then they are a subcategory of The Con Men, aren't they?

I love your "depth, spirit, sweetness, meaning, profundity, truth, and soul" progression, but it also makes me sad. Any audiophile who looks for that in an amplifier rather than in the music itself is, culturally, a pathetic loser, but you are right—that's exactly the way so many of them think. Audio tweakspeak as pornography is, you will recall, a concept that was explored on the cartoon page (1-900-HOT-HIF1) in Issue No. 21, so you know I'm with you there. On the other hand, I'm not with you when it comes to giving up—learning "to roll our eyes and live with the way things are." No way, Howard. Spinoza, schminoza, the simple truth cannot be subverted permanently; it will out in the end. We just have to keep repeating it.

—Ed.

The Audio Critic:

...As you know, I'm a firm believer in the use of double-blind testing, only I use this method with wine, often with interesting results. I'm also personally acquainted with your [Peter Aczel's] honesty, which is impeccable. So when I read that you and your colleagues did not hear differences between CD players, preamps, and amplifiers, I'm certain this is what transpired. I am confused, however.

I also believe in the validity of Bob Carver's transfer-function test (really a way of comparing dynamic performance between dissimilar units) first reported in your journal. If the amplifiers null, then the current and voltage outputs are identical and thus the sound must be as well. There seems to be a basic conflict here: amplifiers that null must sound the same; amplifiers that don't null must sound different, yet I believe you are hearing exactly what you report in your double-blind tests.

In a similar vein, my experience fifteen years ago was that there were substantial differences between amplifiers in their ability to drive real-world loads (leave aside subjective issues of "better"). Two loudspeakers (Gayle and the original Sound Labs) that were both difficult for about six prestigious brands of solid-state amplifiers were handled with aplomb by a Nairn 250, a smaller, much lighter design whose stated purpose was to drive real-world loads and provide plenty of dynamic headroom. I'm not talking about "liquid midrange," but the result was that the Nairn sounded different. I know this is anecdotal, and perhaps amplifiers are much better generally today, but it seems that the ability to drive complex loads will make a difference in the sound, at least under certain circumstances. I am not suggesting anything magical, simply good engineering, and it is a safe bet to say that the Nairn had a different transfer function than the other amplifiers it was compared with.

The transfer functions don't null—different currents and voltages are being delivered to the loudspeaker, therefore they sound different; in double-blind tests no difference is heard—any ideas on how to resolve this dilemma?...

...Best wishes,
Terry McCarthy
New York, NY

No dilemma, Terry. Your puzzlement is based on a false assumption. You write, "amplifiers that null must sound the same; amplifiers that don't null must sound different." Correction: amplifiers that don't null will sound different only if the transfer functions differ greatly. The null test is much more sensitive than the human ear; well-designed amplifiers may be slightly different in transfer function but not enough to be audibly different. Our hearing is relatively crude, so we need to listen to, say, a tube amplifier with high output impedance, lots of second harmonic distortion, and a rolled-off top end before we can distinguish it from a more standard (i.e., neutral) amplifier.

I now have a sneaking suspicion, unprovable after all these years, that the dirty little secret of the original Bob Carver "t-mods" was that the amplifiers sounded indistinguishable from each other even before the transfer-function modifications, at least in the case of solid-state amps. We just didn't have our level-matching procedures down pat. (Sh! Don't tell Bob!)

I am perfectly willing to believe that some of those amplifiers fifteeen years ago had trouble driving crazy impedances—but on easier loads they sounded the same as the Nairn (rhyme unintentional).

—Ed.

The Audio Critic:

I'm an original subscriber to *The Audio Critic* and have enjoyed every issue for the past twenty years or so. *[We don't have quite that much mileage on this vehicle; there was an almost seven-year hiatus in between.—Ed.]* During that time I've noticed many changes in your reviewing perspective, most notably those involving electronic equipment. But while reasons such as double-blind testing have explained some of your changes, there's been one change that I don't believe has been adequately explained. Why do you no longer believe pulse coherence in speakers is audible when you once believed this test revealed the best speakers?

...I may be mixing apples and oranges here, but the only reference I can find that may have led to this apparent shift is cited in Issue No. 16 in speaker reviews by David Rich, where he writes that the "sensitivity of the ear to phase variation remains a point of controversy {Lipshitz 1982], [Fincham 1985], [Deer 1985], [Greenfield 1990]." In a review of the ACI Sapphire II in the same issue, Rich states that the speaker "reproduces pulses with outstanding fidelity" and that the speaker "achieves good pulse response by using first-order crossover sections." Yet, in a more recent review (Issue No. 24) of the updated ACI Sapphire III, Rich writes about the speaker as being "phase-coherent (so the tweaks will respect you)."

Could you please explain this apparent shift in speaker testing and reviewing? Weren't you originally using pulse testing to measure time/phase differences? What were you hearing or measuring in 1977 that you no longer believe you can hear? Do pulse coherence tests no longer show the time-domain differences among speakers that can reveal which are able to make [music sound]... live instead of canned?

Robert Burko
Milwaukee, WI

*Ah, a very good question—good because it is easy to answer. Pulse coherence is intellectually very satisfying, as it indicates waveform accuracy. We all want to believe that the output resembles the input in every respect. Obviously, that's always a good thing, never a fault, as natural to trust as Mom, the flag, and apple pie, so in the early years of **The Audio Critic** it didn't occur to me to question it. The trouble is that we never ran double-blind listening tests to verify our belief.*

The researchers cited by David Rich did run controlled listening tests, however, and pretty much pulled the rug out from under us. The kind of coherence we used to test for—between the midrange driver and the tweeter—is definitely not audible. That's no longer open to argument. At lower frequencies the ear is more sensitive to phase.

The coup de grace to the coherence criterion was actually administered in 1983 by David L. Clark in a not very widely circulated white paper, "Some Experiments with Time" (Syn-Aud-Con Tech Topics, Vol. 10, No. 5, Winter 1983). He reported that at higher frequencies a phase shift of as much as -2700° was inaudible to any of his listeners—i.e., indistinguishable from a wire bypass of the delay network— as long as there was no accompanying frequency response error. (That's the big booby trap, since phase shift will change the frequency response unless the latter is deliberately compensated for.) In 1997 David Clark told me that he has meanwhile carefully trained himself to hear -1000° of phase shift but still gives up on anything between that and 0°!

So, we're back to good old frequency response as the acid test and no longer seek the theoretical comfort of coherence, although we're certainly not against the accurate reproduction of square pulses by a speaker as a techie bonus.

—Ed.

The Audio Critic:

Recently, I was elevated from the post of underpaid prosecutor to that of better-paid judge. Figuring it was time for a few upgrades, I decided to redo my sound system. Plunging into the endeavor with all the enthusiasm of the newly converted, I began reading *Stereophile.*

Now one might think that, because I was a prosecutor for eleven years, I would cast a skeptical eye on the wild claims made by the promoters of High End and tweakdom. Right. I was about as smart a shopper as an eight-year-old set loose in Toys 'R' Us. I bought green paint for my CDs, a "blacklight" to go along with the green paint, and a ludicrous amount of cones, sorbothane thingies, and other useless artifacts. I did everything but bow ritualistically toward Santa Fe and chant, "Harley is God, Harley is God." (Oh yeah, I bought his book. Impenetrable. Now I know why. It doesn't make sense.) Still, there was a wee voice in the back of my mind saying, "Are you out of your mind? A demagnetizer for *CDs?* No use. A junkie is a junkie.

Then I found your magazine at a Denver area newsstand. (Unfortunately, this was on my way back from shelling out $400.00 for new cables.) What a revelation! While it hurt to realize that, yes, I had been suckered by the shills of Santa Fe, I nonetheless felt a certain calm come over me as I read the articles. Sort of like waking from a nightmare—it was very bad, but it's over.

Now, in fairness to Harley et al., I did wind up with a pretty decent system. But I also wound up with various costly pieces of tweaky junk. Were manufacturers in the field of photography (my part-time business) to make the sort of wild claims made by many high-end manufacturers, they would be laughed out of the business. In the real world, companies must make products that achieve identifiable, clearly beneficial results, or they go out of business. They cannot rely on consumers "imagining" they see an improvement, with an assist from hacks with a vested interest in advertising dollars. The thought of anyone trying to market seriously the equivalent of "Shakti stones" or "power line conditioners," in a field where they would have to prove the product's worth, is beyond imagining. Yet folks are getting rich, very rich, by convincing people with too much money that their systems are fatally deficient if they don't purchase some bizarre tweak that in fact does nothing.

Your preference for blind testing and ABX comparisons is so obviously valid that there can only be two possible explanations for Atkinson's resistance to the concept. Either he is afraid of discovering that his precious ears are fooling him (and he doesn't seem that stupid but, then, neither do I) or he knows where the money is and wants it (and the consumer be damned). In the old West, they called it a medicine show. Step up and buy the elixir! It'll cure anything!

There is so much I want to ask you, but rather than take your time (which you don't have anyway), please find enclosed my check for...a new subscription and [all] the available back issues...This is money I don't mind spending.

Also, though I'm not complaining, please keep in mind that not all of us are E.E.s. Some of the technical stuff in Issue No. 24 was way over my head. After all, if I was any good at math and physics, I wouldn't have had to go to law school.

Thanks for your time, and your magazine.

Very truly yours,
Jeffrey S. Ryan
Summit County Court
5th Judicial District of Colorado
Breckenridge, CO

"Joy shall be in heaven over one sinner that repenteth, more than over ninety and nine just persons, which need no repentance." Too bad you didn't discover us sooner. But does it take a law-school education not to treat incontrovertible evidence against tweako cultism with denial? In this delirious hobby it apparently does—and even a law degree isn't always a guarantee, as the number of true-blue audiophool lawyers proves it. As for the E.E. stuff, we are only marginally more technical than Stereophile, *if at all. Some statements about audio are meaningless unless supported with scientific references, but most of our articles are entirely accessible to the interested layman. A few people would like us to be "My First Book of Electricity," explaining what an ohm is (or a µF or a dB) while reviewing CD players and loudspeakers, but that isn't realistic. There are plenty of elementary textbooks in the libraries and bookstores.*

—Ed.

The Audio Critic:

Greetings from the frozen north! Thought I would [share with you some experiences] after years of watching tweakers "modify" circuits. I foil the buggers by potting the circuits of the Morrison preamp. Once saw one of Stew Hegeman's Hapi One phono stages butchered by a wacko. He had replaced all of the parts in the EQ with the usual magic and mysterious components made from yak phlegm. The tolerances were all over the map, but apparently this sort of mundane stuff doesn't matter.

A dealer who was an accountant by trade regularly burned his fingers when "doing the mod" on very expensive amps. One day I watched him solder a film cap with extremely long leads from the negative lug of the large electrolytic on the B+ side to ground. He did the same on the B- side except, of course, from the plus lug to ground. For installing two film caps from ground to ground he charged $400. As later heard via the grapevine, the customer was in seventh heaven over the sonic improvement.

I once sold one of my earlier phono preamps to an outfit called The Tweek Shop somewhere in California. The unit was returned after some time for repair. After lifting the cover I discovered that the unit had been "modified"—parts changed all over the place, complete with melted capacitor casings and several cold-soldered joints. They had the nerve to return the unit, bitching about its performance.

News flash! The audiophile I call Crazy Howard bought $3600 worth of Shakti antivibration-antimagnetic-field-radiation blocks. He wouldn't buy my $800 (Canadian) line stage even though he thought it sounded better than his Jadis but wasn't quite as musical. Uh-huh.

A few years ago a seminar was held on interconnect cables at a Toronto stereo shop. The object of the exercise was to discover if the consumer could pick out the difference between interconnect cable A and interconnect cable B. A room full of keen-eared audio buffs, including a contributing equipment reviewer of *Stereophile,* took part. Unbeknownst to anyone, cables A and B were identical. Those who were objective simply shrugged at the end and confessed that they heard no difference. The neurotics, however—blessed with golden ears and some sort of mystic powers—were able to distinguish subtle differences even when the switch was thrown from cable A to cable A. In other words, when there was no switching between cables at all! Yes, sir! The half dozen or so huddled in a scrum—in the "sweet spot" of course. Furrowed brows, and frantically scribbling notes, and listening soooo hard! This display was a very valuable one. It quickly sorted out the objective listeners from the wackos. And yes, the equipment reviewer was among those who heard things that weren't there. These same people vote in elections.

Following the great interconnect survey, I wanted to investigate further the ability of audiophiles to determine the "sound" of a particular set of interconnects when used in their own system. Two sets of one-meter pairs were fitted with decent locking RCA plugs. One set was covered with blue heat-shrinkable tubing and the second pair with green. Other than colour, both cables were identical.

Once again the objective folks reported that "the goddamn things sound the same to me!" But the rabid audio neurotics, by golly, were not fooled. The differences they heard, though subtle, were unmistakably there and were ranked in terms of "dynamic shadings," "musicality," "coherence," and Christ knows what else. One character wrote a full-page report on the differences between the cables, describing sonic traits never before or since heard of. Remember, the only difference between the two sets was that one was blue and one was green.

Years ago, a friend of mine had the misfortune of assisting a distributor at a hi-fi show. After a boring day of handing out brochures and answering silly "How many watts is that speaker?" sort of questions, my friend decided to stir things up by starting a rumour. It went as follows: whenever a piece of equipment was described, the pronouncement was added, "the umbrella effect is reduced or eliminated." The degree to which it was altered was determined, of course, by careful engineering. Accompanying this claim, the hands were always moved in an arc-like fashion, much like the shape of an umbrella. The person listening to this load of horsefeathers would eagerly nod his head, understanding full well the advantages of taming the dreaded "umbrella effect," and thank heavens here was an outfit at last facing this dilemma head on!

The following morning, in the hotel dining room, there was a sea of assholes arcing their hands in the fashion of das bumbershoot, discussing the advantages of the elimination of the newly discovered source of distortion.

Years ago, I took a stereo nut to his first live symphony. The program was a lively one and included a Borodin piece in which all the players dug in several times in a walloping fortissimo. Despite the fact that we were seated in row five, my colleague announced afterward over a couple of pints that he thought that the live performance would be much louder. This same individual spent his spare time advising other people on how to chose stereo gear...

The Chinese Audiophile Club meets on the first Monday night of the month in Toronto. I was invited as a guest a few years ago by a Mr. Ng to observe the shenanigans that went on. An outfit called Shun Mook (I think) makes discs, pucks, cones, or whatever, out of weird and exotic wood—ebony, rosewood, padauk, purpleheart, etc. The entire meeting that night revolved around the careful listening by the members to the different wooden devices tucked underneath the CD player. The various timber types were actually ranked in terms of being able to "bring out the, best microdynamics" or which one has the "best soundstaging." I suggested that the maple cones, since there was probably a trace of maple syrup, made for a slightly sweeter sound. One of the club members nodded in agreement. I graciously thanked Mr. Ng and beat a hasty retreat. (The calming effect of a Guinness cannot be overrated.)

How come the wackos can hear the differences between blue and green and yet cannot hear a -3 dB rolloff at 10 kHz? How can they hear the difference between oak and ebony cones but cannot grasp the concept of accuracy?

When confronted with virtually distortionless electronics, they always play their trump card. It's not as "musical" as their wretched 10% THD favourites.

Am I pissing in the wind? Should the output of a preamp/power amp not be a replica of the input? Is the "musicality" not the task of Mozart, Bach, and Beethoven?

I could go on for pages but it's just too depressing.

The reason for sending *The Audio Critic* the line stage is to get the idea across that the problem of getting a signal from CD player to power amp is relatively simple, and bog standard parts can be utilized. If just one young audiophile is kept from the clutches of the wackos, it'll make my day!

Regards,
Don Morrison
Morrison Audio
Toronto, Canada

You've made your point, Don. To me the "umbrella effect" alone was worth the price of admission. That I'm also in agreement with your engineering philosophy should be evident from the review of your preamp in this issue.

But tell me, doesn't "bog " mean the toilet in British army and schoolboy slang? I hope "bog standard" isn't something off-colour (to use your bloody spelling, mate).

—Ed.

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To conclude this column, here is a business letter (as distinct from a letter to the Editor) illustrating a fundamental reality of ***The Audio Critic's*** *existence and an important difference between it and other audio magazines.*

Dear Peter:

...I must apologize but we have, with great consideration, decided to refrain from having you review the pieces [Polyfusion Audio CD player and D/A processor] at this time.

Also, again with careful consideration!, we have decided to NOT advertise in your upcoming edition of *The Audio Critic*...

I am sure you are puzzled over this change of heart. This is largely due to our review of some of your statements in the last issue about the amplifier's "vastly exaggerated" importance and CD players and preamps mostly sounding the "same, regardless of price."

We, as a manufacturer of High-End audio gear, obviously disagree with your observations. We certainly respect your right to voice your opinion—but any participation by us with *The Audio Critic* would be non-sensible and non-productive.

Thank you for your understanding.... Perhaps in the future, under different theologies, our paths may cross again.

Sincerely,
Rick Ellis
Sales/Marketing Manager
Polyfusion Audio
Lancaster, NY

What a nice, concise statement of the high-end audio industry's unshakable belief that the truth is bad for business!

A number of important corrections are in order. To wit:

Our statements regarding the sound of purely electronic signal paths are not mere "observations" or "opinions" or "theology. " They are reiterations of incontrovertible facts of electrical engineering and psychoacoustics as attested by authorities with the highest scientific credentials.

Secondly, sound is not the only reason to buy high-end audio electronics. To use an automotive analogy, a Chevrolet will get you to the mall as quickly and reliably as a Bentley, but that perception hasn't killed too many Bentley sales.

Lastly, an advertiser wishing to be "productive " needs a publication with a demographically optimal readership, not with a groupie philosophy. Take it from an old Madison Avenue professional.

—Ed.

The Good Guys in the White Hats and the Bad Guys in the Black Hats: Continued

By Peter Aczel
Editor and Publisher

You will need Issue No. 24 to begin this expanding list of good and bad audio people at the beginning and understand it as completely as intended.

The article about the White Hats and Black Hats of audio drew by far the strongest reaction, from both our friends and enemies, among all the items in our last issue. It appears that the prurient interest exceeds the technical and musical interest even within our very special circle of readers. (I knew it all along.) Nominations for additional white and black Stetsons kept pouring in, but most of them were not white enough or black enough to qualify, being various shades of gray. (See also page 14, bottom right.) As an example, take someone like *Bob Stuart* of Meridian Audio. He is without doubt a gifted, scholarly, and highly credible technologist, one of the pillars of the audio engineering community. Yet he has also been the high priest of a contingent of English tweaks who hear things that don't exist. He is certainly not a Black Hat in the Bruce Brisson sense, but I have qualms about placing him in the White Hat category. And so it goes.

Be that as it may, there were clearly some omissions in the original list. Here are a few names that should have been included—and, of course, sequels are to be expected.

More White Hats

It should be pointed out that the White Hats are generally not as visible as the Black Hats. You have to look for good things that are happening in audio and find out who is behind them. Unlike most of the Black Hats, these good guys are often modest and don't promote themselves.

Paul Barton *(PSB Speakers)*

Another Canadian. (They have disproportionate representation on the White Hat roster, possibly because the snake-oil tradition is not as deep-rooted in their mercantile culture as in ours.) Paul Barton's compatriot John Ötvös, maker of my reference speaker (Waveform Mach 17), once told me that the PSB Stratus Gold would be his choice for his own use among other manufacturers' speakers. That's good enough for me, although I hope to test one soon in our laboratory. In any event, Paul has been known to me for years as an absolutely straight, scientific, nontweako, truth-telling engineer, devoted to integrity in speaker design. That gets him a White Hat.

James Bongiorno *(Spread Spectrum Technologies, Inc.)*

The badass boy wonder, now aging, of the electronic engineering community. Those of our readers who go back to the '70s probably remember my various run-ins with Jim, but that doesn't change the fact that he is a highly original and ceaselessly creative circuit designer, not only in audio but also in RF. Those long-ago SAE, GAS, and Sumo topologies Jim had cooked up turned out to be classics, very different from the expected and clearly superior. His patented but never used FM detector circuit is truly innovative. ("If someone actually manufactured this tuner, it would rule the world," says David Rich.) Jim is currently unaffiliated (the company name above is that of his consulting firm) but he has some big plans. He recently sent me a long, rambling, and barely legible manuscript (totally illegible to my scanner software) that I hesitate to publish because it withholds the main piece of information that would make it truly interesting and important. Jim claims that the diff-amp (differential amplifier) input in solid-state ampifier design is antiquated, passe, obsolete, and that he has a more advanced solution. What precisely? He won't say; he just teases us with some hints. (He probably thinks the plagiarists are lying in wait, salivating.) The thing is—I believe him because I believe *in* him. He is one of the few audio people capable of advancing the art and not just shooting the bull (at which he is also very good).

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Editor's Note: Some time after the above was written, there came the news that Jim Bongiorno was very sick. He has constantly recurring and now life-threatening liver problems that go back many years, and he needs some very sophisticated and costly medical treatment for which he has no insurance. Eventually he may have to undergo a liver transplant, which is contingent on organ availability as well as funds. We are talking about six-figure *sums here, totally beyond Jim's means. Contributions are being solicited to The James Bongiorno Medical Trust Fund, Bank of Montecito, Santa Barbara, CA 93140. The contact for the fund is Ms. Krissi Wray at (805) 564-0220.*

Fred E. Davis *(independent consultant)*

There have really been only two sane, scientifically modulated voices amidst the strident voodoo chants on the subject of wires/cables: that of Dick Greiner, already designated a White Hat in the last issue, and that of Fred Davis (in the AES *Journal,* in *Audio,* and elsewhere). Modesty prevents me from naming a third voice—never mind. Fred Davis tunes the RLC differences and their audibility/inaudibility even finer than the rest of us rationalists but leaves just as much egg on the voodooists' faces. Good man, good exegete.

Bill Dudleston *(Legacy Audio)*

The lantern of Diogenes would have revealed him to be an honest audio designer. His top-of-the-line loudspeaker systems (not yet tested here) are very highly regarded by all the experts I respect and, best of all, cost about half of what they would if distributed through dealers, thanks to Legacy's factory-to-consumer marketing. What's more, he is a straight talker on all technical matters. Good engineering, good value, and no B.S. add up to a White Hat in my book. (I have charitably decided that the very few and far-between "eyebrow raisers" in the Legacy literature must originate from someone else in the company.)

David Griesinger *(Lexicon)*

The man who knows more about digital signal processing for multichannel sound than just about anyone else. You could say he wrote the book. He is the engineering community's spokesman on the subject; they look up to him, and so do I. Ask him about psychoacoustics, too; he'll tell it like it is. Also read the Lexicon DC-1 review in this issue.

Tomlinson Holman *(TMH Corp., formerly Lucasfilm)*

A very serious audio technologist by anyone's reckoning. Remember the Apt/Holman power amp of the early '80s? It was the smartest, most completely thought-through design of the era (and I didn't even fully appreciate it at the time). Fast-forward to THX, into the later '80s and right up to the present time. That was entirely Tom Holman's doing, and even if the THX system has certain promotional/commercial undertones it has undeniably helped to stabilize the surround-sound scene and set a minimum performance standard. Do you like the sound of the Indiana Jones movies? Tom Holman was in charge of the entire technical infrastructure there. Anyone who has attended a few Audio Engineering Society conventions can testify to Tom Holman's intellectual leadership. The world of audio would be quite different without him.

Dr. Roger West *(Sound Lab, Inc.)*

Definitely the most credible of the electrostatic loudspeaker gurus. He has never denied the disadvantages of the electrostatic design approach; he just believes that the advantages are decisive. His top-of-the-line Sound Lab Ultimate 1 overcomes the disadvantages through brute force but still doesn't carry a cynically inflated price tag; his smaller models are honestly engineered and honestly priced tradeoffs. David Rich is a big fan of the entry-level Quantum; I myself have tested the next step up, the Dynastat, but want to look at a more current production version before publishing a review. In my book, Roger West is "Mr. Electrostatic USA" on account of his scientific integrity.

More Black Hats

If this is the part you started to read first, you have a dirty mind and are directed to go back to the White Hats and start there. It is with considerable sadness that I list the names below—or do you think I just want to give you a cheap little frisson of *Schadenfreude?*

George Cardas *(Cardas Audio)*

Worse than Bruce Brisson? That's how some snake-oil monitors rate him. He is certainly a major example of the breed—the cable peddlers who use pseudoscience to justify their insanely inflated prices. Cardas cable is designed with "Golden Section Stranding"—strands differing in mass in accordance with a Fibonacci sequence or "Golden Ratio"—in order to eliminate "resonant multiples in the conductor." This is such garbage in terms of real-world audio engineering that Pyramid Power is the pinnacle of applied science by comparison. Look up www.cardas.com on the Web and laugh your head off. Or cry.

Martin Colloms *(Stereophile, HFN/RR)*

A particularly opprobrious Black Hat because he possesses sufficient technical knowledge and scientific logic to be totally aware of what he is doing to his gullible readers—filling their heads with tweako garbage. He has the ability to perform all the laboratory tests this publication does, and then some, after which he will report that cable A has much better rhythm and pace than cable B and that some loony single-ended triode amplifier with oodles of distortion is the cat's meow. His hypocrisy quotient is right up there with fellow Brit John Atkinson's, and so is the technocultural damage he does.

Martin DeWulf *(Bound for Sound)*

Another audio writer/editor opposed to the findings of practitioners who, unlike him, are trained in science. Of course, one can never be sure whether such marginal pundits are sincere or not. Unless he keeps claiming to hear differences that in reality do not exist, the expensive toys he likes to play with and write about might stop coming. He has nothing to offer a high-end manufacturer except his exquisite auditory aperçus and facile audio-salon prattle, since he lacks the technical wherewithal to come up with constructive criticism in engineering matters. His limitations lock him into a groupie position. A guest article by DeWulf in a recent issue of the *The Absolute Sound,* listed as a "think piece" in the table of contents, rehashes without any evidence of thinking all the tired old sounds-the-same/no-it-doesn't issues that have been laid to rest, over and over again, by some of the most highly qualified experts in audio and psychoacoustics. My name appears in the article, as well as in a followup exchange of letters in a later issue, as a centerpiece of DeWulf's demonology. He cannot forgive my long-ago defection from the subjective reviewing of purely electronic signal paths. What he doesn't seem to understand is that some of us are capable of gaining new knowledge and new insights over the years, not just weight.

Michael Green *(RoomTune by Michael Green Designs)*

The room-treatment charlatan extraordinaire. His only saving grace is that, if you fall for his cockamamie theories and buy his gizmos, you are out less than $1K, in most cases. Does that make him less of a Black Hat? I don't think so; antiscience is antiscience, regardless of the price tag. Some of his little Velcro-fastened dinguses aren't big enough to affect the acoustics of an airplane toilet, let alone the sound of a big listening room. Sure, when he hands them out free of charge to exhibitors at a show, you see them used in the rooms and you think they are there for a purpose, but think again—who says no to a freebie? Especially when it takes up little room, is basically harmless, and comes from an affable chap with Jesus hair? (All that hair will fit only into a size XL black hat.)

Benjamin Piazza *(Shakti Audio Innovations)*

Designer and promoter of the Shakti Stone, a.k.a. Shakti Electromagnetic Stabilizer, a mind-boggling example of the tweako artifact that does nothing and of the gullibility of the high-end audio freak. A Shakti Stone is a magic brick, $199.99 each (but you must buy several), which you simply place on top of an audio component to defeat electromagnetic interference (EMI) and thereby improve the sound. Whatever is inside the inextricably potted brick—who knows, maybe it's all kinds of passive circuitry as Piazza claims in his technobabbling "white paper" or maybe it's bat guano—that's not how it's done by scientifically accountable engineers even where EMI is a problem (as it rarely is in audio). You've seen this before—the bold leap from an esoteric but technically defensible buzzword to a fatuous marketing non sequitur. *Question:* What's the difference between a Shakti Stone and a Shun Mook Mpingo Disc? *Answer:* Who cares?

Jonathan Scull *(Stereophile)*

The audio journalist who brought tweako subjective reviewing to a new height of unbridled self-indulgence and foppish silliness. I think even John Atkinson must feel slightly nauseated as he edits JS's fulsome copy. It is significant that the most outrageously tweaky products that come in for review are usually assigned to JS. If they ever reviewed a liquid-nitrogen-cooled platinum power cord, JS would be the reviewer, and the name of his wife Kathleen would be invoked to corroborate his exquisite sonic perceptions. To me he represents the ultimate in hokum, self-congratulation, and unaccountability.

Dr. Yu Wah Tan *(Shun Mook Audio, Inc.)*

Assisted by *Bill Ying* and *Andy Chow*—and I don't know (and don't care) which member of the trio is the originator of the steer droppings they hype. Maybe it all comes from a secretly located stockyard in Chinatown. All I know is that "Mpingo" (ebony) wood pucks placed in strategic spots on and under your audio equipment don't accomplish anything other than fattening the early-retirement fund of the Shun Mook phonies. Actually, anyone so stupid as to spend money for these worthless New Age fetishes deserves what he is getting. Thus, in the final analysis I'm all for Shun Mook because their efforts help to identify and isolate the irredeemable idiots in our midst. (See also Don Morrison's letter in the "Box 978" column in this issue.)

The Academy Advancing High End Audio and Video

Not an individual but an alliance of manufacturers, most of whom make their living with overpriced tweako products. I say most, not all, because paradoxically there is a small sprinkling of White Hats among them, strictly for self-serving political reasons, in the face of fundamental differences in engineering doctrine. As an organization, however, they stand for all the antiscientific voodoo and subjectivistic rubbish that this publication fights against. Their members banded together to lobby for the audibility of the inaudible. They definitely don't want you to know that bits are just bits, that wire is just wire, that vacuum tubes are horse-and-buggy devices, that magic objects placed on top of an amplifier can't make it sound better. They want you to be a True Believer. The Stetson factory never made enough black hats for all the heads that are screwed on wrong in this self-styled "academy."

A New Category: Gray Hats

This is a kind of purgatory for seemingly contrite Black Hats on their way to possible redemption.

Corey Greenberg *(Audio, Stereo Review)*

When I classified Corey as a Black Hat in the last issue, I pointed out his potential for rehabilitation in an improved intellectual environment. I swear he must have listened to me because at Hachette he is definitely cooling it with the coolo lingo and unbridled subjectivism. He is doing good work. That pearl gray Stetson is *you,* Corey.

How to Be a Sophisticated Audiophile and Resist Trendy Stupidities

By Peter Aczel
Editor and Publisher

Are you a new reader of our publication? Or a regular reader looking for more let's-get-down-to-basics information? This is addressed to you (but it won't hurt the more advanced aficionado, either).

Serious thinking about any subject begins with the irreducible fundamentals. Why would any sane individual want to get involved in the finer points of audio technology, i.e., be an audiophile? What is the purpose of it all?

The obvious purpose is music on demand in the home, with a sound quality as close to the real thing as possible. The trouble is that the obvious purpose of anything is not necessarily what is served by its actual implementations. For example, the obvious purpose of a very expensive restaurant is great food, but there are some that serve so-so food and are frequented mainly for lifestyle affirmation and celebrity watching. By the same token, there are people who don't really like music but own $200,000 sound systems. I wouldn't know how to advise such people, although they sometimes ask me.

Anyone about to spend a significant amount of money on audio equipment should, as a first step, just stop and think for a few days. Where are you headed? What do you hope to accomplish with your purchase? If you are planning to listen to a lot of music and want it to sound as good as possible, that's one thing. If you want to turn a large room in your house into a movie theater, that's another thing (although a dual-purpose solution is not out of the question). If the orthopedic surgeon down the road owns a Ferrari and you want to one-up him with your new audio equipment, that again is a different ballgame (which I won't help you play). Whatever your goal, you must be able to define it, otherwise you will be all over the place in your buying decisions and probably end up frustrated. That goes for techie types as well as novices.

Here I am going to assume that you love music, own or plan to own a lot of recordings, know what live music sounds like, and have no special audiophile agenda. (I shall deal with the more common agendas, dogmas, fetishes, and manias, but I know you are unlikely to listen to me if you have already been co-opted.) The highest form of audiophilia is knowing how to obtain optimum results in the listening room without any concern for the audio-salon ideologies of the moment.

There. I said "room." That's where the serious thinking must begin. If your room is small, you must lower your sights. Reproduced sound of great textural and timbral refinement is achievable in a small room, but a truly life-sized, spacious, authoritative sound with untrammeled dynamics is not—not even if you throw money at it. (Years ago, Harry Pearson of *The Absolute Sound*—and now also *Fi*—became the laughingstock of knowledgeable observers when he flaunted the original, monstrously large and expensive Infinity IRS loudspeaker system in a phone-booth-sized room in his Long Island house.) Not that a large room guarantees good results. If your room is one of those architectural prize winners with huge undraped picture windows, a bare terrazzo floor, and sparse furniture, you can expect dreadful sound in it even if it is large and has a cathedral ceiling. For the best possible sound, you need a room that is both large and well-damped. How to treat a room for optimum damping is a basic question beyond the scope of this article, but there is no dearth of information on the subject in books and magazines. Here I merely want to emphasize that "think room" should be the watchword of the enlightened audiophile, whether starting from scratch or just planning improvements.

About Loudspeaker Systems

If you believe your listening room is all it can be, the loudspeaker system is what you must think about next—and most carefully. Don't let any tweako/weirdo audio cultist tell you otherwise (even if he has a byline in a magazine or owns an audio store): *the loudspeaker, in combination with the room, will determine the overall sound quality of your system, not the electronics.* If you are vague or cavalier about your choice of speaker and terribly intense about the amplifier and other electronic components, you will be joining the great brainwashed who pour their money down the sinkhole of a fantasy market. If the only thing you remember after reading this article is that the loudspeaker *rules*, my effort will not have been wasted.

Unfortunately, there are very few genuinely accurate speaker systems at any price, let alone under $1000 or even $2000 the pair. By accurate I mean that the output of the speaker resembles its input to a high degree, over a reasonable solid angle in front of the speaker. If that seems like an oversimplification, you will find more detailed and more finely differentiated explanations in the loudspeaker section of nearly every issue of ***The Audio Critic.*** One way to plan the purchase of an accurate speaker system at a less than scary price is to give up the deepest bass temporarily, settle for a superior *small* or *medium-sized* loudspeaker model initially, then acquire a high-quality subwoofer (or two) at a later date. This is the crucial crossroads in system planning or upgrading; it is very easy to mess up here and set out in the wrong direction. On the other hand, once you have the right speakers, it is very difficult to screw up the rest of the system because honest values in electronics are readily obtainable today; your choices are far from limited.

Pitfalls to Avoid

This is a basic orientation article, not an equipment survey, so I am not going to make specific speaker recommendations here, but I do want to point out certain common pitfalls and obfuscations you are likely to encounter in your search for the right speaker system. First of all, beware of ultrahigh-priced speaker cables. That whole industry is a fraud. There is no reason to pay more than a dollar a foot, and even that is probably overkill. (For example, good commercial-grade RG-8 coax cable, which is reasonably low in both inductance and capacitance and can be used with the center wire and the shield as the two loudspeaker conductors, costs 69 cents per foot at Radio Shack.) I recommend that you read the wire/cable article in Issue No. 16; it will keep you out of the clutches of the cultists and charlatans.

You will also avoid paying a double price, for whatever speaker cable you might choose, if you eschew the biwiring fallacy. Even enlightened audiophiles fall for that one, despite the fact that it defies one of the basic laws of physics, the superposition principle. As it relates to electronics, the superposition theorem states that any number of voltages applied simultaneously to a linear network will result in a current which is the exact sum of the currents that would result if the voltages were applied individually. If you believe in science, you cannot possibly believe in the biwiring ritual. Note that I am talking about biwiring, not biamping. (The latter is not without justification in some instances, although its full benefits are obtainable only with a system driven from a line-level electronic crossover—meaning no passive network connected to the drivers.) Note further that biwiring is quite harmless even though nonsensical. That otherwise reasonable loudspeaker manufacturers recommend it shows how intimidated they are by the prevailing tweako market forces.

What kind of crossover design is best is another subject replete with tweako booby traps. The first-order cultists will tell you that only 6-dB-per-octave (first-order) crossover slopes are any good because they sum to a coherent waveform and can pass square waves. The trouble with that theoretical advantage is that (1) it is only true at the "sweet spot" and (2) you can't hear it, as David Clark and others have demonstrated over and over again. High-order slopes, on the other hand, reduce distortion in the stopband, permit better power handling, and in most cases yield a less "lobey" response off axis. My experience has been that 24-dB-per-octave Linkwitz-Riley crossovers are the best (with the possible exception of Rich Modafferi's Infinite Slope configuration, still in very limited use) and that electronic crossover units are preferable to passive networks. In any event, do not opt for a speaker just because it has a first-order crossover and no other compelling reason.

There is also the matter of speaker stands. Small speakers need to be raised off the floor to listening height, no doubt about that, but the stands need to be merely solid and stable, not necessarily gyrostabilizer platforms. (I exaggerate only slightly considering the techie-phony designs out there that sell for big bucks.) The big spikes that are de rigueur on high-end stands will prevent rocking on a thick carpet but on a bare floor they are as useful as teats on a bull (not to mention lethal to polished wood flooring). Be sensible in your selection of speaker stands; there is nothing more there than meets the eye.

One more word of warning about loudspeakers. Be aware of the shortcomings of unconventional transducer technologies, i.e., other than electrodynamic direct radiators. Some electrostatic loudspeakers offer very beautiful sound, but only the largest (such as Dr. Roger West's two or three biggest Sound Lab models) have adequate power handling at realistic playback levels with dynamic program material. Horns handle power with aplomb but with rare exceptions have highly colored response. As for sci-fi solutions, such as ionized air, acoustical heterodyning, rigid sheets, etc., you are on your own and should expect no encouragement from me, although it is conceivable that one of today's insufficiently debugged exotic designs is the wave of the future.

As for multichannel setups, the ground rules still apply. A good loudspeaker is a good loudspeaker, whether it handles your front left, rear right, or center channel.

With a good subwoofer in a 5.1 system, the bass capabilities of the surround speakers are not all that critical, even though in the Dolby Digital (AC-3) mode all five speakers receive a full-range feed. Some compromise in size and bass extension is practical and reasonable for the center and rear channels in most cases, but the up-to-date approach is to avoid dinky little satellite-type speakers altogether. Should a speaker design be optimized differently for home theater than for music? I hold with those who feel that the question is more political (or ecclesiastical?) than practical, but I am not about to question the Tom Holman doctrine when it comes to a THX-calibrated home theater for movies only. Ask me again when every home has a dedicated media room. For now, my criterion remains the sound of music, not the localization of dialogue or suchlike.

Fact and Fiction in Audio Electronics

Willy-nilly, we come to the electronics from which the speakers are fed. I put it that way because I dread the unavoidable debate about vacuum tubes versus solid state. It is not an intellectually respectable debate in the late 1990s, any more than (let us say) typewriting versus word processing. To have to sermonize on the subject is a professional embarrassment. As I have said many times before, there is nothing wrong with correctly designed vacuum-tube equipment if you already own it; it will in all likelihood sound just fine (unless it was deliberately designed to tweak the signal rather than to reproduce it accurately). But—to go out today, in the golden age of silicon, and spend big bucks on new vacuum-tube equipment is the height of folly. If the tube equipment happens to be a single-ended triode amplifier, then folly is too weak a word; idiocy would be more appropriate. Anything that vacuum tubes can do, solid-state devices can do better, more reliably, and at lower cost. Even the deficiencies of vacuum tubes, such as relatively high second harmonic distortion, can be mimicked by solid-state circuitry if the designer happens to like the euphonious coloration that results.

When it comes to choosing solid-state electronics, another debate rears its muddled head. Does a high-end (i.e., Krell, Mark Levinson, Spectral, etc.) power amplifier or preamplifier sound better than a typical midpriced (i.e., Pioneer, Sony, Yamaha, etc.) unit? The educated answer is—why should it? The midpriced equipment also has high input impedance, low output impedance, flat frequency response, low distortion, and low noise—and that is what we can hear. There is no such thing as an effect without a cause, and there is nothing to cause the high-end equipment to sound better. Needless to say, I would rather have a Krell than a Pioneer as a birthday present, for reasons that have nothing to do with the sound. What reasons? Better build quality, greater reliability, more beautiful appearance, better retention of value, greater pride of ownership, more attentive treatment by the company in case help is needed—should I go on?

I am convinced that the myths of one amplifier "blowing away" another in a side-by-side listening test are mostly due to the difficulty of matching levels within ±0.1 dB. It is a fussy and boring process that tries your patience, even if you have the proper equipment to do it with. When the levels differ by as little as 0.25 dB, there is an audible difference, which will be immediately interpreted by some audiophiles as a blow-away. Of course, there are those who can clearly hear a difference in an A/A test as long as they think it is an A/B test. (I have tried that dirty trick on a number of 'philes.) Trust me, no one has ever, ever distinguished two properly designed amplifiers or preamps by their sound alone in a valid blind test. So—the question to ask is not how the equipment sounds but how it meets your goals and satifies your needs. You need a more elaborate array of controls on your preamp if you have a complex sound system than if you have an extremely simple one. You need more watts out of your power amp in a large listening room than in a small one. You need better build quality if you change your system every ten years than if you change it twice a year. And so forth. It is basically common sense.

Are there any fictions, cults, manias, and fads to guard against when choosing solid-state audio components? Nothing as grotesque as the single-ended triode craze, but certain fahionable buzzwords should automatically activate your B.S. warning light. (Just the warning light, not necessarily the B.S. shutoff.) For example, *discrete* circuitry isn't necessarily better than high-quality op-amps. *FETs* aren't necessarily better than, or even as good as, bipolar tansistors for a given application. *Class A* isn't necessarily an indication of superior quality. *Low feedback*, or *zero feedback*, is often a less desirable design approach than high feedback, correctly applied. *Polypropylene* capacitors do not "sound better" than less costly capacitors, correctly used. (See also the article on high-end prejudices in Issue No. 24, pp. 16-23.) My pet peeve is the word *speed* when applied to audio electronics. Those who use it think they are being very cool, very professional-sounding, but actually they betray their ignorance. All electronic signal paths that are dead flat to, say, 22 kHz have unlimited "speed" (which in my book is just another way of saying "bandwidth") for audio purposes. I defy you to distinguish supposedly "fast" and "slow" amplifiers from one another in a blind test.

When it comes to multichannel AV electronics, there are genuine differences, and not only in watts per channel—that's the easy part. The control circuits range from ultrasophisticated digital, with highly advanced processor chips, down to yesterday's leftover analog with minimal digital implementation. Here you need expert help, which is generally not available from the dealers and certainly not from the tweako magazines. We have

(continued on page 40)

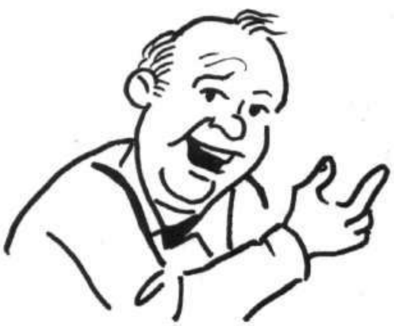
In Your Ear

I'm one of the old-time audiophiles.

Back in the vacuum-tube days, I used to pull my tubes and have them checked on a tube tester every three or four weeks.

One of my tube amps had front-panel bias adjustment—/ used to fuss with it for hours.

Then came solid state and spoiled all that.



But we still had those great tone arms that let you change the vertical tracking angle. I used to fuss with that for hours. And I kept the stylus clean with a tiny sable brush.

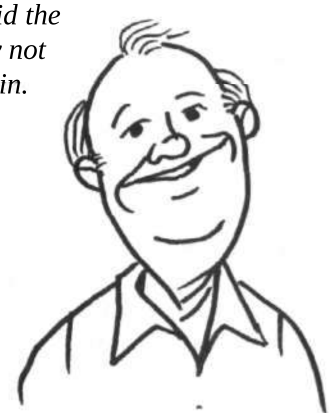
Then came CD and spoiled all that.

I was happy when surround sound came because I could fuss for weeks with speaker placement. But then my wife put her foot down and said the room will absolutely not be rearranged again.

My life was empty. I had nothing to do but listen to the damn music.

Then I bought one of those single-ended triode amplifiers the audiophile magazines are raving about.

Now I pull my tubes and have them tested every three or four weeks. I put them back and have the service guy check the bias.



The tube revival saved my life. I'm a born-again audiophile.

About Minimonitors, Subwoofers, and Full-Range Systems In Between

By Peter Aczel
Editor and Publisher

Size is what mainly determines how a speaker system is designed, and there are always many possible engineering solutions in every size category. Here is a sampling of current design practice.

Our readers must be reminded, and should understand, that we cannot go back to square one in every issue and explain our philosophy of loudspeaker evaluation over and over again. Ideally, you should have a complete set of our publication beginning with Issue No. 16. If that is not practical, go back at least four or five issues if you are a new reader. Not that there is anything in our procedures that radically departs from established scientific practice, but we are constantly bombarded with questions that have already been answered in our pages. (Back issues are available at the same rates as new subscriptions.)

Here I want to add that the various sophisticated computer programs currently available for loudspeaker design—crossovers, woofer enclosures, simulated system response, etc.—have virtually eliminated truly god-awful speakers. There appears to be a slow convergence toward relative accuracy, although one still runs into some willful and often self-contradictory design tradeoffs.

There has also been a significant improvement in the raw (OEM) drivers available to loudspeaker manufacturers from the various American, European, and Asian raw-driver houses. Nasty ringing in response to tone bursts has become so rare that I no longer do tone-burst tests routinely, only when I suspect energy storage problems. Advances in materials technology and CAD must be credited for this. Frequency response and distortion specs are also better across the board, even in mass-produced drivers. One more reason for the aforesaid convergence.

Bag End Infrasub-18

Bag End Loudspeakers, Modular Sound Systems, Inc., P.O. Box 488, Barrington, IL 60011. Voice: (847) 382-4550. Fax: (847) 382-4551. E-mail: info@bagend.com. Web: www.bagend.com. Infrasub-18 powered ELF-system subwoofer, $1495.00. Tested sample on loan from manufacturer.

In Issues No. 21 and No. 22, David Rich and I covered in considerable detail the ins and outs of the Bag End ELF system and the design of Bag End subwoofers. At that time Bag End was a brand widely recognized in the field of professional sound but not in the audiophile world. Since then the company has been exploring new directions and has developed a product called Infrasub-18 for the high-end consumer market.

The Infrasub-18 rolls the Bag End concepts into a single integrated package. The unit incoporates an 18-inch driver in a 3-cubic-foot sealed box, a 400-watt power amplifier, and a somewhat simplified version of the ELF electronics. The power amplifier is something new; the pro-sound users had to supply their own. Its 400-watt continuous power rating is presumably into *4Ω.,* which is the nominal impedance of the EL-18P driver. The built-in ELF module does not have to be as flexible as the separate pro-sound version, as it is dedicated to a single subwoofer. At the early 1997 introductory price of $1295.00, this package represented outstanding value. With the $200.00 price increase, the Infrasub-18 is now positioned at the same price point as the top-of-the-line Paradigm product, and the criteria inevitably change.

I do not intend to explain the ELF system all over again in this review. For that you will have to go back to Issues No. 21 and No. 22. Here I only need to note that the ELF "concealment" system causes the f_3 (the -3 dB bass cutoff corner) to slide up and down the frequency scale as the power into the driver goes up and down. The frequency response of the Infrasub-18 was dead flat (±0.5 dB) down to 10 Hz, the limit of my measurement capability, as long as I didn't push the 1-meter SPL into the high 90s (as normalized to 40 Hz). That's flatter and deeper than anything else in the business; the Infrasub-18 is unique in that respect. (The printed claim is actually an f_3 of 8 Hz.) Theoretically, the resulting improvement in group delay across a significant portion of the low-frequency band should have some audible effect, but I can't confirm that. The total subjective impact of a subwoofer depends not only on low-frequency extension but also on distortion, damping, maximum SPL, etc., so I would need to set up an ABX test between two Infrasub-18's, a standard one and a modified one with a much higher small-signal f_3 but otherwise indentical, in order to zero in on that single sonic characteristic.

Speaking of distortion, that's not where the Infrasub-18 shines. The FFT spectrum of a 20 Hz tone at a 1-meter SPL of 90 dB (into 2π steradians, i.e., out on the open floor) shows the 2nd harmonic to be at -20 dB (10.0%), the 3rd harmonic at -22 dB (7.9%), the 4th harmonic at -33.5 dB (2.1%), the rest negligible. Not very impressive, since 90 dB is a fairly modest SPL where the ELF concealment circuit does not even come into play at 20 Hz. The 30 Hz distortion at the same SPL is -23.5 dB (6.7%) 2nd harmonic, -38 dB (1.3%) 3rd harmonic, the rest negligible. Going up to 40 Hz at the same SPL, which should really be a breeze, the 2nd harmonic is still -26 dB (5.0%) and the 3rd harmonic -49 dB (0.35%). Raising the 40 Hz SPL to 97 dB, which is far from unreasonable, the 2nd harmonic rises to -22.5 dB (7.5%) and the 3rd harmonic to -42 dB (0.8%). I also took a THD sweep from 100 Hz down to 20 Hz, at a 1-meter SPL of 91 dB as normalized to 50 Hz. That was still below the concealment threshold as far as I could tell. The THD curve breaks sharply at 70 Hz, where the distortion is only 0.21%, and rises in an almost straight line against the logarithmic vertical scale to 13% at 20 Hz, more or less confirming the in-between frequencies as measured in the FFT tests.

Thus, the Infrasub-18 appears to be the powered subwoofer with the most extended low-frequency response but also the highest distortion, at least among the designs known to me. Is there an audible downside to that? None that I could discern at the SPLs I am able to tolerate in my listening room. The sound of the Infrasub-18 is basically as impressive as I described in my original review of the Bag End S18E-C with ELF-1 outboard electronics. With the simplified built-in electronics you connect your signal source to the line-level input and your main amplifier to the highpass outputs, set the level, and you're in business. There's a polarity switch if you need it for the most seamless crossover (I did). The crossover slopes are inherently 12 dB per octave with the ELF system, and the crossover frequency is factory-set at 95 Hz on the Infrasub-18. It all works very smoothly. The ease of matching the sub to the main speakers is arguably greater than with other systems. If this were the only supersub on the market, I could live with it happily forever after.

It so happens, however, that several other high-performance powered subwoofers are significantly lower in distortion and have greater SPL capability per dollar. At its original introductory price, the Infrasub-18 would still be a "best buy." At its current price, I would like to see better distortion figures. This journal still regards the ELF system as just another interesting and plausible engineering approach, not a breakthrough. Our Dr. David Rich had an extended dialogue with designer Ron Wickersham and ended up respectfully disagreeing with the latter that the ELF bass-boost circuit is an "integrator." No matter how you slice it, it's still an equalizer. (A classic case of a rose by another name that smells as sweet.)

Bag End MM-8B Time-Align

Bag End Loudspeakers, Modular Sound Systems, Inc., P.O. Box 488, Barrington, 1L 60011. Voice: (847) 382-4550. Fax: (847) 382-4551. E-mail: info@bagend.com. Web: www.bagend.com. MM-8B Time-Align nearfield monitor, $2264.00 the pair. Tested samples on loan from manufacturer.

I have always found that the downside of coaxial driver design is much greater than the claimed advantages, undeniable as they are. The heart of this small monitor for the professional market is an 8-inch woofer with a horn tweeter firing through the apex of the cone. The 1.75-inch mouth of the horn is extended by the flare of the woofer cone. The transition between the two drivers is not smooth, as it hardly ever is in a coaxial configuration, and there are serious disturbances in frequency response as a result. This inherent design problem can be solved by making the woofer and tweeter both coaxial and coplanar, as in the now defunct Win SM-10, which had a proprietary Japanese frame that could accommodate flat diaphragms in the same plane. No other solution really works in my experience.

The Bag End MM-8B monitor incorporates Ed Long's familiar Time-Align technique, which goes back to 1976 and is claimed to yield superior performance in the time domain. Indeed, the acoustical output of the speaker shows an unusual degree of time coherence, with the leading and trailing edges of square waves still quite well preserved after passing through the electroacoustic transducer. I attach relatively little importance to that these days, although I used to; I am convinced now that you can't hear it. "Look, Ma, no phase shift" is intellectually appealing but perceptually insignificant; David L. Clark proved conclusively in 1983 that the phase shift has to be gigantic before it becomes audible. (Except at low frequencies, but that's a horse of another *écurie.)* I didn't have my head set straight on this subject until much later, and there are readers who still keep reminding me of my former allegiance to time-coherent design. Hey, I'm still not against it; like chicken soup, it can't hurt; but the coherence mustn't be obtained at the expense of frequency response, as it is here.

The MM-8B varies by ±7 dB in MLS-derived axial response. If one ignores a huge 11 kHz suckout (14 dB from peak to trough), the response is still no better than ±5 dB. There are three EQ switch positions for brightness, affecting the response only above 6 kHz; they change the level but do not correct the roughness. The speaker is intended to be used as a nearfield studio monitor and (not surprisingly) sounds quite aggressive, near and far, in all switch positions. Efficiency is very high, of the order of 93 dB SPL, 1 W, 1 m. The compression-type tweeter, crossed over at 2.9 kHz, and the slot-loaded bass system, with an f_3 I measured at 90 Hz (!), are both design features with efficiency as the top priority.

What about distortion? Very low at typical listening levels, thanks to the high efficiency. At a 1-meter SPL of approximately 90 dB, I found the nearfield distortion of the system fluctuating between 0.14% and 0.5% from 210 Hz up and no worse than about 1% from that frequency down to the f_3.I did not take measurements at higher levels because I was satisfied that I had seen the general trend. There is a whole litany of specifications printed on a sticker affixed to the back panel of the speaker, and one of them is "less than 1% THD 100 Hz to 20 kHz 94 dB spl @ 1 meter." I think that's pretty much on the money.

The impedance of the MM-8B does not dip below 8Ω at any frequency but rises to *45Ω.* in the vicinity of the crossover. The phase angle fluctuates a great deal but stays within ±45°. No special amplifier requirements are indicated.

What all this adds up to is punchy and clean but rather highly colored sound and very little bass. For some specific in-your-face monitor applications in the studio, maybe on top of the console, that may be just the ticket, but for the audiophile the MM-8B has little to offer at an uncomfortably high price. Well, maybe it offers a simple moral: when everything is right except the frequency response, the sound will still not be right.

Hsu Research HRSW12Va

Hsu Research, Inc., 3160 East La Palma Avenue, Unit D, Anaheim, CA 92806. Voice: (714) 666-9260. Fax: (714) 666-9261. E-mail: hsures@earthlink.net. Web: www.hsuresearch.com. HRSW12Va powered subwoofer with 250-watt amplifier option, $1000.00 each (factory-direct, including shipping/handling). Tested sample on loan from manufacturer.

There was a last-minute "Flash!" at the end of the HRSW12V review in Issue No. 24 commenting on the (then) new "a" mod. Here is the promised detailed update, belated and quite possibly outdated, as the model nears retirement while still being highly typical of the Hsu line.

The improvements in the driver and port design have not changed the previously reported distortion curves more than I would expect my somewhat crude measurements to vary in two different sessions on two different occasions. This remains one of the lowest-distortion subwoofer designs without motional feedback. At the lowest frequencies (20 Hz to 30 Hz) the distortion is quite comparable to that of the Paradigm Servo-15a, for example, though not in the more sensitive 50 Hz to 100 Hz octave. Even so, a nonservo sub that never rises above 5% to 6% THD at any frequency as you push it to the limit is quite exceptional.

The HRSW12Va box (or rather cylinder) is tuned to about 17 Hz, yielding a measured f_3 (-3 dB point in the small-signal response) of the same frequency. The overall frequency response is within ±1 dB from 80 Hz down. The outboard amplifier is a bit more powerful and a lot more conventional-looking than its predecessor; the new driver can easily handle the extra power; and the flared ducts have pretty much eliminated the low-frequency chuffing at high SPLs—but overall this is still the same excellent subwoofer as before. At $1K it's no longer a huge bargain like Dr. Poh Ser Hsu's original cardboard barrels of years ago, but show me a powered sub that's better or even as good at that price. There aren't any.

JosephAudio RM7si "Signature"

JosephAudio, 2 Pineridge Road, White Plains, NY 10603. Voice: (800) 474-4434. Fax: (212) 724-2509. E-mail: josephaud @aol.com. *Web: www.josephaudio.com. Model RM7si "Signature" 2-way minimonitor loudspeaker system, $1699.00 the pair. Tested samples on loan from manufacturer.*

This model is the improved or deluxe version of the RM7si that was so enthusiastically reviewed in Issue No. 24. As Voltaire said, the best is the enemy of the good. The plain-vanilla RM7si, good as it is, has been eclipsed by the Signature version, which now rises to the rank of the best minimonitor-sized speaker system known to this publication.

How much better is the Signature than the basic RM7si? Well, it sounds more open, more detailed, better balanced, more neutral overall. The difference is not huge but any experienced audiophile can easily hear it, and at $400 extra anything less than that would be cause for rejection.

What is new in the Signature? Mainly the drivers, which are still the same size (1-inch dome, 6½-inch cone) but not the same design. The new woofer has an aluminum, rather than fiberglass, diaphragm. The vented box has not changed. Whether the proprietary "Infinite Slope" crossover had to be modified to match the new drivers I do not know. The fact is that you have to examine the speaker quite thoroughly before you can see it is the Signature version.

The frequency response of this version is very flat. The small "step" in the response of the basic RM7si (see Issue No. 24) is gone. Except for two small dips at 2.7 kHz and 5.3 kHz, each 3 dB or so, the 1-meter MLS measurement on the Signature's tweeter axis yields an almost ruler-flat curve. On the woofer axis the curve remains almost identical, with just the slightest broadening of the first dip. That "Infinite Slope" crossover is certainly doing its job. It has made the measurement axis quite uncritical compared with other designs, even 4th order. The low-frequency response of the Signature version indicates a very slight upward creep of the f_3, which may or may not be merely the difference between two quite similarly tuned boxes measured on two different occasions. The vented box of the Signature appears to be tuned to 42 Hz, which is also the tuning frequency of the basic RM7si (give or take a hertz), but the summed response of woofer and vent is -3 dB at 45 Hz rather than 43 Hz as in the basic model, and there is also a slight bump of a little over 1 dB at 73 Hz, whereas the basic model is Butterworth-flat. If I had a dirty mind I'd suspect the new-and-different-driver-in-the-same-old-box syndrome, but then the impedance curve has hardly changed, and besides Rich Modafferi knows what he is doing, so I'll stay in the clean-thinking modality. This is still outstanding bass response considering the size of the driver and of the box.

As for distortion, my comments in Issue No. 24 still apply. Below 100 Hz the Signature is slightly better, above 100 Hz ever-so-slightly worse, but it's basically the same profile. The significant improvement is not in distortion but in frequency response above 1 kHz. That's what we hear.

One thing that should have been emphasized a little more in the original review is that the ingenuity of the Infinite Slope concept lies in the way it solves *a passive* crossover design problem; active (electronic) crossovers can be designed with very steep slopes quite conventionally. I believe, with many others, that powered loudspeaker systems are the wave of the future, so the significance of Rich Modafferi's superb engineering solution may diminish in the years to come. For the moment, it rules—not even the truly excellent Paradigm Active/20 powered speaker equals the RM7si Signature in a side by side comparison. You do have to hitch it to an amplifier, though.

Legacy Audio "Studio"

Legacy Audio, 3023 East Sangamon Avenue, Springfield, IL 62702. Voice: (800) 283-4644 or (217) 544-3178. Fax: (217) 544-1483. E-mail: legacy@fgi.net. Web: www.legacy-audio.com. Studio compact monitor loudspeaker, $948.00 the pair (direct from Legacy). Tested samples on loan from manufacturer.

When you buy a loudspeaker from a dealer, you know he gets at least 50 points from the manufacturer, maybe more. By cutting out the dealer and selling directly to you, Legacy is in effect offering you a speaker that competes in the $1000-per-side retail class, or thereabouts, for $474 per side. The appearance of the Studio certainly confirms such a classification; build quality and finish are truly superior. When an enclosure takes up only 0.8 cu. ft. of space but weighs 30 lb., you know it's built like a brick outhouse.

The speaker is designed around a 7½-inch bass/midrange driver (Kevlar cone, dual voice coil) and a 1-inch titanium-dome tweeter (a departure from Legacy's ubiquitous ribbon for the topmost frequencies). The crossover frequency is 2.8 kHz (lst-order lowpass, 2nd-order highpass, each with some frills). The 13-by-ll-by-10-inch box is vented to the rear. A toggle switch shelves the low bass for use with a subwoofer; another toggle switch allows two different tweeter levels.

My MLS (quasi-anechoic) measurements yielded mixed results. The 1-meter response of the system on the axis of the tweeter is almost amplifier-flat up to 3.5 kHz but very rough further up. Of course, the flat part of the curve is in the 7½-inch driver's range, even if measured on the tweeter axis; the roughness is all tweeter. Hunting for the "sweet spot" improves the tweeter response to ±3 dB (but no flatter) all the way to 20 kHz. There is a huge suckout at around 15 kHz, no matter where the measurement is taken, and a roller coaster profile in the 3.5 kHz to 7 kHz octave. At 30° off axis that same octave sags, whereas the 7 kHz to 12 kHz response remains strong. Strange. The front baffle is not very tightly wrapped around the two drivers; there's quite a bit of "land," and that plus the sharp edges may well be the culprit. The tweeter level switch has only a 1-dB range, so its effect is minimal.

The nearfield responses of the woofer and vent add up to a 40 Hz box, give or take a hertz, all nice and flat. That's as good as it gets, considering the size of the driver and the enclosed volume. The impedance of the system fluctuates between 4.3Ω and 20Ω in magnitude (4Ω nominal) and ±45° in phase—not the easiest load for an amplifier but far from strenuous.

This being a small system, I took distortion measurements only at a 1-meter SPL of approximately 91 dB, normalized to 5 kHz for the tweeter and 500 Hz for the bass/midrange driver. That's loud but not brutal. The tweeter has absolutely negligible distortion in its range, typically 0.1%; the bass/midrange fluctuates between 0.1% and 0.9% down to 92 Hz; at 50 Hz it's still only 2%. That's an excellent result.

How does the sound of the Studio relate to its measured performance? Quite neatly. The flat, low-distortion bass sounds very solid and musical. You will not feel deprived even if you don't have a subwoofer. Further up the sonic texture is a little coarser than with, say, the JosephAudio RM7si Signature; that last bit of refinement and airy transparency is lacking; however, there is no significant coloration, as the small roughnesses are random, not tending toward a specific frequency band. Those who hate a "hot" top end will like this speaker because it never sizzles, not even when the program material is sizzly. The overall personality of the speaker suggests comfortable musicality rather than the highest accuracy. I could live with the Studio and be reasonably happy, but I rate a number of small speakers in this survey higher.

Monitor Audio 700PMC

Monitor Audio USA, P.O. Box 1355, Buffalo, NY 14205-1355. Voice: (905) 428-2800. Fax: (905) 428-0004. E-mail: goldinfo© monitoraudio.com. Web: www.monitoraudio.com. Model 700PMC 2-way minimonitor loudspeaker system, $999.00 the pair. Tested samples on loan from manufacturer.

The new Monitor Audio PMC series, of which this is the lowest-priced model, features woofer cones made of a special material called ceralloy, colored in gold, and gold-anodized tweeter domes. I don't think I'm being influenced by the gilded look when I say that this is a very fine loudspeaker, especially in view of its diminutive size. Among the bookshelf-type speakers reviewed here, only the JosephAudio RM7si Signature and the Paradigm Reference Active/20 can be said to exceed it in transparency, lack of coloration, and overall sonic credibility—and both of those are bigger and costlier. The question is whether or not $1K is still too much for an unpowered minimonitor that absolutely needs a subwoofer for music with more than minimal bass and dynamic range.

The 700PMC is designed around a 6¼-inch woofer with the ceralloy cone and a 1 -inch tweeter with the gold-anodized metal dome. There is barely room for the two drivers on the front baffle, necessitating a rearward-firing port for the vented box. The crossover, not specified in the literature that I have, appears to be at 3.3 kHz with first-order slopes.

The frequency response is very smooth and varies relatively little between the two driver axes. I obtained the flattest 1-meter MLS response on the tweeter axis, where it remained within ±2 dB from 300 Hz up to 20 kHz. There is a shallow two-octave trough centering on the crossover frequency and a slight decline above 13 kHz, both of which may have contributed to the strictly nit-picky attack/transparency/delicacy margin between the 700PMC and my top choices. At 45° off the tweeter axis there is still strong response up 11 kHz and a falloff above that frequency. The phase response barely varies in slope on either driver axis. Very nice results, overall.

The bass response is of course less impressive. The vented box is tuned to 64 Hz, and that's the "corner" of the LF profile. Nothing wrong with that in a minimonitor, but for a bass addict like me it leaves something to be desired. As for distortion, I measured it only at a 1-meter SPL of 90 dB (normalized to 5 kHz for the tweeter and 500 Hz for the woofer), since minimonitors are not required to rattle the windows. The tweeter at that level fluctuated between 0.06% and 0.4% THD from 2 kHz to 20 kHz, remaining below 0.1% between 3.3 kHz and 12 kHz. That's nothing short of superb. The woofer remained below 0.5% THD down to about 150 Hz, but at the lowest frequencies the distortion rose to quite high levels, as expected. Good design is clearly evidenced by all of these figures.

The impedance of the 700PMC represents an easy load for the amplifier, never dropping below 7 ohms in magnitude and staying within approximately ±30° in phase. The tuned box kicks up to 20 ohms on the impedance peaks.

Bottom line: it's small, cute, well designed, far from cheap, and sounds very good indeed.

NHT Model 2.5i

Now Hear This, Inc., 535 Getty Court, Building A, Benicia, CA 94510. Voice: (707) 747-3300. Fax: (707) 747-1252. Customer service: (800) NHT-9993. Web: www.nhthifi.com. Model 2.5i floor-standing 3-way loudspeaker system, $1300.00 the pair. Tested samples on loan from manufacturer.

The NHT Model 3.3, Ken Kantor's "statement" loudspeaker, was featured on the cover of Issue No. 21 and enthusiastically reviewed, at considerable length, in that issue. Model 2.5i is essentially a scaled-down embodiment of the same design principles, so I see no need to go over the same ground again in this review. At less than one third the price of the illustrious flagship, the 2.5i cannot be expected to deliver comparable performance—and it doesn't. It is merely an excellent speaker.

Unlike the 4-way Model 3.3, the 2.5i is a 3-way system, with a sideways-firing inboard 8-inch woofer, a 6V2-inch polypropylene midrange, and a 1-inch aluminum dome tweeter. The two smaller drivers are on the narrow inboard-angled (21°) front baffle, for "Focused Image Geometry" (see the review referred to above). The woofer enclosure is vented in the back, so the speaker cannot be placed against the wall like the sealed-box Model 3.3. The crossover frequencies are 100 Hz (2nd order lowpass, 1st order highpass) and 3.3 kHz (2nd order highpass and lowpass). The basic look is that of a mini-3.3, down to the stabilizer bars for the bottom of the narrow cabinet.

You will recall, if you have read Issue No. 21, that the 3.3 was incredibly flat in frequency response and low in distortion. The 2.5i is no slouch in those respects but not quite up there. Its relatively flattest 1-meter response is on the tweeter axis perpendicular to the plane of the front baffle, where I measured ±2.5 dB from 300 Hz to 20 kHz. On the midrange axis the response is only slightly worse, ±3.5 dB between the same limits. It should be noted that the most abrupt ripples are in the 2 to 9 kHz range, where the ear is sensitive. The top octave of the tweeter is almost ruler-flat. The straight-ahead response (i.e., on the cabinet axis, 21° outboard from the drivers) is still pretty flat except for an 8 dB suckout at the crossover frequency. The bass response, as measured at the best summing junction of woofer and vent, is also nearly ruler-flat down to 28 Hz (the vented box is tuned to 30 Hz). Of course, with an 8-inch woofer, the penalty for such deep bass is low efficiency. I made no precise determination of the latter but estimate it to be no better than 83 dB/1 W/lm in the woofer's range. Impedance fluctuates between 3.6Ω. and 10Ω in magnitude (6Ω. nominal) and between ±30° in phase—an extremely easy load for any amplifier.

I measured THD only at a 1-meter SPL of 90 dB. At that fairly high but still not stressful level the nearfield distortion of the summed woofer and vent hovered around the 1% mark at all frequencies down to 25 Hz, where it began a rapid climb to 10% at 20 Hz. The midrange distortion remained between 0.2% and 0.7% at all frequencies except for a somewhat strange excursion to 2.5% at 1.2 kHz, where there is no apparent resonance or other explanation. The tweeter never exceeded 0.1% THD in its range. Overall, these are very good distortion figures, almost in a class with the 3.3, except of course in the bass.

When it came to the subjective evaluation of the Model 2.5i's sound, I was somewhat disappointed. Not because the 2.5i doesn't sound like a first-rate speaker; it does. I just expected more, on the basis of the above measurements and the speaker's heritage. I expected not only neutrality (i.e., low coloration), which the speaker possesses in spades, but also a bit more refinement of detail, transparency, authority, "this-is-it-ness." Maybe I should have lived with the speaker a little longer. I wish I could be less vague (not my style, as you know), but there it is. Even so, a sophisticated low-coloration speaker with deep bass in a handsome enclosure at $650 per side is nothing to be sneezed at.

Paradigm Reference Active/20

In Canada: Paradigm Electronics Inc., 101 Hanlan Road, Woodbridge, ON L4L 3P5. Voice: (905) 850-2889. Fax: (905) 850-2960. In the U.S.: AudioStream, Division of Bavan Corporation, MPO Box 2410, Niagara Falls, NY 14302. Voice: (905) 632-0180. Fax: (905) 632-0183. Web: www.paradigm.ca. Reference Active/20 powered 2-way bookshelf loudspeaker system, $1600.00 the pair. Tested samples on loan from distributor.

I have believed, and proclaimed, for years that the power amplifier and the loudspeaker are in effect a single system, the back end of the audio chain, and should ideally be designed as an integrated unit. The market, on the other hand, has steadily resisted such an approach, at least until recently. (Audiophile orthodoxy requires an a la carte choice of each separate component, and especially of the power amplifier, otherwise those exquisite judgments regarding the "speed" or "graininess" or "layering" of bipolar versus MOSFET output stages, not to mention single-ended triodes, would remain unexercised.) Today, the penetration of the home-theater market by powered subwoofers is making it easier for powered main speakers like the Paradigm Reference Active/20 to gain acceptance. It is plain common sense that a lean-and-mean dedicated amplifier is a more efficient engineering solution than a fat all-purpose amplifier, but common sense is not what usually prevails in the delirious world of high-end audio.

All of the above would be little more than enlightened theorizing were it not for the outstanding performance of the Active/20. This product represents contemporary audio engineering at its best. Exactly how good is it? So good that, if it were only a *tiny* bit better, the ultra-high-end loudspeaker business would be in big, big trouble. Luckily for the megabuck speaker makers, the Paradigm stops just a hairsbreadth short of "ultimate" performance. But it's close, uncomfortaby close.

Not that the Active/20 isn't high-end. This publication, unlike certain others, doesn't consider $1.6K to be mid-fi. But look what you get for $800 per side. A very solidly built, bookshelf-size, thoroughly dead enclosure, available in a choice of attractive finishes; a 6½-inch bass/midrange driver with a high-tech mica-loaded polymer cone and die-cast basket; a 1-inch aluminum-dome tweeter, also with die-cast chassis; two built-in power amplifiers (110 watts for the bass/midrange, 50 watts for the tweeter); an electronic crossover (3rd order, 1.5 kHz); a full complement of calibrated controls; choice of RCA or XLR (balanced) input; and an extra long AudioStream shielded interconnect with RCA plugs (tweako style, B.S. arrows printed on the insulation to show signal direction—but, hey, it's harmless, it's sturdy, and it's free). That's pretty good value, and all of it is quite nicely built. (One exception: the heavy toroidal power transformer of the internal electronics could be more securely mounted; it came loose in one sample I looked at.)

Obviously, I couldn't have such a high opinion of the Active/20 if the measurements weren't good. But they are. The 1-meter frequency response on axis is very flat, ±2 dB, up to about 14 kHz; above that it rises to an 18 kHz peak which flattens out off axis. Whether the axis is considered to be that of the woofer or of the tweeter is quite uncritical; the curves are almost the same. At 45° off axis the response is still excellent, ±3 dB all the way up to 18 kHz. The low-frequency response is necessarily limited but very respectable; the vented box with its rearward-firing ducted port is tuned to 42 Hz, which is the -3 dB point of the 4th-order Butterworth response profile. A highpass filter (3rd order, 100 Hz) is activated by a toggle switch to roll off the bass for use with a subwoofer.

At a 1-meter SPL of 90 dB, normalized to 400 Hz, the nearfield THD sweep of the woofer started at 0.15% at the top of its range, remained at that level down to 300

Hz, then rose steadily to a maximum of 3.6% at 32 Hz. The tweeter at the same 1-meter SPL, normalized to 6 kHz, fluctuated between 0.05% and 0.15% THD over its operating range, pretty much like other 1-inch dome tweeters. This admittedly limited exploration of the Active/20's distortion performance indicates perfectly well-behaved drivers, unlikely to be pushed beyond their capability by their dedicated amplifiers.

The sound quality resulting from these performance characteristics is very high, as I have already stated. The speaker is not quite as neutral, transparent, and subtly refined in sonic detail as the JosephAudio RM7si "Signature" (see above) or, especially, our reference Waveform Mach 17 (see below) but it definitely nudges the borders of that category. Only by quickly switching to those superior speakers (as driven by external electronics) is the difference clearly perceived. Considering the advantages, in cost and convenience, of the Active/20's internal electronics, the difference may not be important to the prospective purchaser. This is a highly persuasive piece of audio equipment, capable of giving satisfaction to demanding audiophiles, and it can expand into a self-powered super system by the addition of the Paradigm Servo-15a subwoofer, whose review follows.

Paradigm Reference Servo-15a

In Canada: Paradigm Electronics Inc., 101 Hanlan Road, Woodbridge, ON L4L 3P5. Voice: (905) 850-2889. Fax: (905) 850-2960. In the U.S.: AudioStream, Division of Bavan Corporation, MPO Box 2410, Niagara Falls, NY 14302. Voice: (905) 632-0180. Fax: (905) 632-0183. Web: www.paradigm.ca. Reference Servo-15a powered subwoofer, $1500.00 (including X-30 crossover/control unit). Tested sample on loan from distributor.

I find it a little difficult to pinpoint the place of this unit in the high-end subwoofer pecking order. I would rank it higher than the Bag End Infrasub-18 but lower than the Velodyne Servo FSR-18—if it were that simple, which it isn't. For example, the Bag End is easier to match to the main speakers, the Paradigm sub is the obvious bass extender for their Active/20's, and the Velodyne is relatively overpriced without offering greater low-frequency extension. And so on. The truth is in the details.

The Servo-15a is a motional-feedback subwoofer designed around a 15-inch driver, a 400-watt internal power amplifier, and an outboard crossover/control unit (X-30). Its external dimensions are 20 in. (height) by 18 in. (width) by 22½ in. (depth). The X-30 controls the phase, lowpass cutoff frequency, and level of the sub, while providing highpass outputs for the main speakers with a choice of three crossover frequencies. Thus an elaborate AV front end (such as the Lexicon DC-1) would eliminate the need for the X-30, but in most installations some, or all, of the functions of the latter will be used. Other powered subwoofers usually have these functions consolidated on the back panel.

Are the sealed-box loading, the power amp, and the accelerometer-controlled servo loop of the Paradigm fundamentally different from the design used by Velodyne? Without blueprints and circuit schematics I can't have an answer to that question, but I discern somewhat different design priorities on the part of the Paradigm engineers. The external evidence of that is the relatively higher THD of the Paradigm—but still much lower than the distortion in open-loop systems—accompanied by low-frequency performance quite comparable, and even superior, to that of 18-inch subs. What could be the reason for such a difference? The design of the driver, for one thing. The design of the accelerometer, for another, not to mention the placement of the accelerometer. Maybe all three.

The measured performance of our Servo-15a sample was outstandingly good. The small-signal frequency response was ±0.75 dB down to 21 Hz and -3 dB at 18 Hz. That is very similar, almost identical, to the response of the Velodyne Servo FSR-18 (see below). A nearfield THD sweep of the sub at a 1-meter SPL of 100 dB (normalized to 50 Hz) resulted in a curve that starts with 0.15% at 110 Hz, crosses 0.2% at 52 Hz, begins to break sharply at 44 Hz, crosses 1.0% at 30 Hz, and rises to 4.1 % at 20 Hz. The nearfield FFT spectrum of a 20 Hz tone, with the 1-meter SPL set to 95 dB at *that* frequency, showed the 2nd harmonic to be at -26 dB (5.0%), the 3rd harmonic at -32 dB (2.5%), the rest negligible. Thus the Paradigm distorts a lot less than ordinary woofers but quite a bit more than the Velodyne.

Then there is the question of maximum SPL capability. At 20 Hz, the rapidly rising distortion in the vicinity of 100 dB suggests that there is little margin left before the limiter clamps down. (All powered subs have some kind of built-in limiting.) Above 40 Hz there appears to be considerably more headroom; I estimate that limiting would take place only past 110 dB—but I am not willing to expose my ears, my shelves, and my windows to testing at such levels, as I am not really interested in sound tracks with car crashes and rocket launches. In any event, the Servo-15a is at least as capable SPL-wise as any 15-inch sub known to me.

As far as subjective listening quality is concerned, the Paradigm is right up there with the very best. Organ, bass drum, double basses, etc., are reproduced with stunning realism, and the relatively higher distortion vis-a-vis the Velodyne Servo FSR-18 is not audible when playing music (as distinct from test signals), unless more prolonged listening than I have done proves otherwise. This is one of the best powered subwoofers money can buy, and that's $899 less money than the price of the Velodyne.

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The combination of the **Paradigm Reference Servo-15a** with a **Paradigm Reference Active/20** pair (see the front cover of this issue) is a $3100 stereo ampliher/speaker system whose performance would be hard to beat with conventional equipment costing three times as much or more. If your budget for the "back end" of a stereo system is in that ballpark (and, remember, the 2.1 combination is readily expandable to a cost-effective 5.1 surround-sound array), I cannot recommend anything else more highly. The only thing you risk is losing the love of power-amp manufacturers.

A word of caution if you take this route. Correctly level-matching the Active/20's and the Servo-15a to one another is a fussy, time-consuming affair, even if you follow the instructions faithfully. I strongly recommend one of the inexpensive Radio Shack sound level meters and some CDs with frequency sweeps, pink noise, and warble tones to do the job. You'll be frustrated if you try to do it by ear. I fine-tuned the combination very successfully and ended up with splendid sound, but it took a while. Considering the sound quality per dollar, the effort is well rewarded.

Pinnacle Classic Gold Aerogel

Pinnacle Loudspeakers, 101 Commercial Street, Plainview, NY 11803. Voice: (516) 576-9052. Fax: (516) 576-0826. E-mail: pinnacle@pinnacle-speakers.com. Web: www.pinnacle-speakers.com. Classic Gold Aerogel Tower 3-way loudspeaker system, $1695.00 the pair. Tested samples on loan from manufacturer.

This is the new top-of-the-line full-range model in the Pinnacle line, which starts in the two hundreds and includes all kinds of speakers—stereo, home theater, in-wall, subwoofers, etc. It is a vented system, with three fat diagonally slanting tubes called Diaduct acting as ports in the back, and four drivers: two 8-inch woofers with paper cones and rubber surrounds, a 5¼inch midrange with the so-called Aerogel treatment, and a 1-inch ferrofluid-cooled metal-dome tweeter. The crossover frequencies are 500 Hz and 5 kHz; the network is quite complex, with filter slopes and shelving circuits tailored to the drivers, and built with high-end components such as air-core coils and polypropylene capacitors. The speaker is rather tall (44 in.), quite narrow *(9½* in.), and of moderate depth (15 in.).

One of the most difficult tasks a speaker designer can set for himself is to engineer a monolithic system with a small footprint that does it all, covering the full audio spectrum with high efficiency and low distortion, while keeping the cost within reason. It's a delicate balancing act, which the Pinnacle engineering team (headed by Richard Rothenberg and Peter Moore) has executed about as neatly as I've ever seen. This is a genuinely excellent all-round loudspeaker system a discriminating audiophile can live with and not go into debt. I was impressed.

The frequency response of the unit is best measured on the tweeter axis, as it is less good on the midrange axis. Interestingly enough, the 1-meter response improves in the stereo position, 30° inboard. In the 0° position the tweeter level is set too high, an average of 3 dB above the midrange level. The midrange response itself is very flat up to about 3 kHz, and the tweeter response is also reasonably flat up to 14 kHz, but there is this distinct upward step in the 3 to 4 kHz band, which looks much better 30° off axis. In the latter position I would characterize the response as ±2 dB right up to 16 kHz, except for a 5 dB dip at 3 kHz. The low-frequency response, as aided by the Diaduct ports, appears to be quite flat down to 26 Hz but is hard to measure accurately because of the opposite-firing woofers and ports. The vented box is tuned to approximately 35 Hz, maybe a little lower.

I measured the distortion of the system only at the 1-meter SPL of 90 dB. At that rather loud level the tweeter is virtually distortionless, fluctuating between 0.06% and 0.13% THD. The midrange is also clean, 0.07% to 0.5%. The woofers, naturally, distort a little more, 0.2% to 1%. I think these THD figures are outstandingly low. Needless to say, the speaker can play a lot louder than 90 dB; it is also more efficient than most.

The impedance curve of the system fluctuates between 3.4Ω. and 15Ω. in magnitude (8Ω nominal) and ±40° in phase, with smaller swings than that over most of the audio range. Just about any decent amplifier can drive such a load.

The sound resulting from the sum total of the above characteristics is essentially neutral, highly defined, transparent, and dynamic. The bass is particularly excellent; no subwoofer is needed, certainly not for music. If I wanted to nitpick, I would perhaps point out a very slight heaviness in the lower midrange—or is it the upper bass? It may be due to floor reflections but is in any case quite minor, not enough to trigger an investigation. Bottom line: if a tall, floor-space-saving, full-range speaker is what you need in your room and the price is in the ballpark for you, I can unreservedly recommend the (what a mouthful!) Pinnacle Classic Gold Aerogel Tower.

Sunfire "True Sub Signature"

Sunfire Corporation, 5210 Bickford Avenue, Snohomish, WA 98290 or P.O. Box 1589, Snohomish, WA 98291-1589. Voice: (425) 335-4748. Fax: (425) 335-4746. Web: www. sunfire labs. com. "True Subwoofer Signature" 2700-watt powered subwoofer, $1895.00 each. Tested sample on loan from manufacturer.

Bob Carver was never afraid of banalities in advertising and sales promotion, so he labels his top-of-the-line products with the hackneyed and ubiquitous "Signature" designation and silk-screens his John Hancock on the control panel. That tells you the Signature sub is a step up from the original "True Subwoofer," and with a 46% higher price tag it had better be. My immediate reaction to it was that I wished Bob had launched his breakthrough idea in this version, as the THD is so much lower, the SPL capability even greater, and the size only slightly larger—a 13-inch cube instead of the original 11-incher, i.e., still extremely compact. It is fairly obvious that the 13-inch version costs very little more to manufacture, the parts being either the same or very similar, so the original launch price could have been the same or close to the same. Marketing, marketing...

We ran almost four pages on the original True Subwoofer in Issue No. 24, so I am restricting this review to the differences found in the Signature. There are very few. Size is the most obvious one, the 13-inch cube housing a 12-inch (o.d.) driver and a 12-inch (o.d.) passive radiator. They are scaled-up copies of the original 10-inch versions. The electronics I found essentially indistiguishable from the original; there are probably some subtle internal differences beyond my ken.

One major change: the family of lowpass filter curves controlled by the "Crossover Freq." knob has been radically shifted toward the lower frequencies. The silkscreen calibrations of 30 Hz to 100 Hz are entirely incorrect; the "100 Hz" curve is in fact -18 dB at 100 Hz (relative to 45 Hz, where the flat response is centered) and -10 dB at 80 Hz (the surround-sound subwoofer reference frequency). That makes the Signature unmatchable to small satellite speakers, but according to Bob Carver nobody wants to use it that way. He claims the feedback from dealers and customers indicates bottom-end extension of full-range speakers to be the desired use. The instructions enclosed with the unit do not reflect that—not in the case of our early sample, anyway.

Frequency response is sufficiently similar to that of the 11-inch Sunfire to require no new discussion. I measured a slightly higher tuning frequency (22 Hz) for the passive-radiator-vented box, but that could easily drift downward with additional break-in of the surrounds. What has clearly been improved is the distortion. Even when radiating into a half sphere, by far the most stringent measurement condition, the Signature can produce a relatively low-distortion 20 Hz tone at a 1-meter SPL of 93 dB. A THD reading taken right off the passive radiator (the active driver is virtually silent at that frequency) was a very respectable 4%, in a league with high-end subwoofers of much larger dimensions. Raising the frequency to 50 Hz and the SPL to 100 dB resulted in a THD reading a little under 1% (this time off the active driver, the passive radiator's contribution being negligible at that frequency). The 40 Hz distortion at 100 dB was a little harder to measure because of a small residual contribution by the passive radiator; my estimate was 2%. It would appear that increasing the size of the True Subwoofer by the ratio of $13^3/11^3$ was all that was needed to bring the THD into the range represented by some of the better "big mother" units. (One peculiarity: line-in L produces 8 dB more acoustical output than line-in R—with the same input! Maybe Bob thinks L-R = 0 is bad for car crashes.)

As for maximum SPL capability, I have explained elsewhere why I don't run extensive tests at triple-digit dB levels. Extrapolating from the relatively low 100 dB distortion, I would expect levels in excess of 110 dB to be achievable without limiting at all but the lowest frequencies.

One little boo-boo of the Signature should be pointed out. I was unable to eliminate a slight but audible ground-loop hum no matter what I tried—floating the line-cord plug with a ground-breaking adapter, reversing the polarity of the plug with the aid of a nonpolarized adapter, grounding the back plate to the ground of the input device, and all permutations and combinations of the above. I measured the best-case 60 Hz hum to be at a 1-meter SPL of 57 dB, so the music will always cover it up; even so, there appears to be a downside to the transformerless power supply.

Bottom line: the Signature is clearly superior to the original True Subwoofer, which was an astonishing piece of engineering to begin with, and which both David Rich and I regarded with considerable awe at its debut. The upgraded design's substantial virtues and minor limitations as discussed above should pretty much define its market; for certain audiophiles it will undoubtedly be the powered subwoofer of choice.

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Reviewer's Note: *The Sunfire "True Subwoofer Signature " is quite new and arrived here late in the game, long after the Velodyne Servo HGS-10 review below was written. There was no time to rewrite the latter and reexamine the Sunfire/Velodyne comparison, especially since Velodyne also makes the HGS-12, a cube comparable in size to the Signature and not yet tested by us.*

Thiel SCS2

Thiel Audio, 1026 Nandino Boulevard, Lexington, KY 40511. Voice: (606) 254-9427. Fax: (606) 254-0075. E-mail: mail @thielaudio.com. Web: www.thielaudio.com. Model SCS2 coaxial 2-way bookshelf loudspeaker system, $1990.00 the pair. Tested samples on loan from manufacturer.

Reviewer's Note: *The following was written long before the SCS2 was replaced by the SCS3 ($2600.00 the pair!), which is so new that you are unlikely to have seen a review of it anywhere. The SCS2 review is retained here because (1) I think it is still highly instructive and (2) it provides a probable reason for the short shelf life of the speaker. The basic architecture of the SCS3 appears to be the same.*

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This one I still can't believe. As our regular readers know, I don't entirely agree with Jim Thiel's design philosophy and priorities, but I have never had my hands on a Thiel speaker that didn't have ruler-flat frequency response. Until now. Initially I thought I was testing a defective sample but then I found both speakers in the pair to be almost identical, so that excuse was eliminated.

I refer you to the Bag End MM-8B review above. Same situation here—coaxial design creates more problems than it solves (unless the drivers are also coplanar). The 1-inch aluminum-dome tweeter of the SCS2 fires through the apex of the 6½-inch woofer's aluminum cone and runs into serious interference by the latter. Curiously, the Thiel literature explains that other ("typical") coaxial speakers behave like that but no, not theirs, as the woofer diaphragm is shaped to prevent interference.

Well, my quasi-anechoic (MLS) measurements, taken at 1 meter on the speaker's perpendicular axis, show incredible roughness, swinging within an 18 dB range from 1 kHz to 20 kHz in the better sample and a 25 dB range in the worse sample. The biggest dips are at 3.3 kHz and 8 kHz, the biggest peaks at 4.6 kHz and 12.5 kHz. Moving 30° off axis (the stereo position) helps a little above 4 kHz, but the roughness is still dramatic. The literature shows a ±2 dB axial response, so who's crazy here? John Atkinson's measurements as published in *Stereophile Guide to Home Theater* some time ago appear to bear me out. The dips and peaks in his curve are a little bit less severe than in mine, but the overall profile is virtually identical. Of course, Wes Phillips's subjective evaluation of the SCS2 in the same review is nonetheless extremely positive—Thiel is one of the sacred cows of the high-end community and mustn't be blasphemed. I, on the other hand, thought the speaker sounded quite colored, maybe not as hopelessly colored as the measurements suggested but basically noncompetitive in its price range.

The root cause of the problem is—you guessed it—Jim Thiel's *coherence-über-alles* doctrine. Coaxial speakers are relatively easy to make coherent, and the SCS2 passes square pulses quite nicely, even with its un-Thielish 2nd-order crossover. (Could it perhaps be the Ashley-Kaminsky "quasi-2nd-order" network of the early '70s? That would fit the picture, but the SCS2 literature doesn't say.) Coherence at the expense of frequency response, however, is a terrible tradeoff—optimizing the inaudible while degrading the audible. I refer you once again to the Bag End MM-8B review above for a more complete explanation. (I hate to repeat myself.)

The good features of the SCS2? Beautiful build quality. Quick convertibility from quasi-sealed-box to vented-box design. (You pull the foam plugs from the unit's two ducted ports; the f_3 moves from 80 Hz down to 42 Hz; the bass response remains flat, but the efficiency drops—very neat trick considering Jim had to work with one and the same driver Q.) Low woofer distortion down to well below 100 Hz. (It's difficult to measure distortion above 1 kHz, since the level varies all over the place.) Provision for wall mounting. Hey, a design with the wrong priorities can still have quality touches.

I don't like to write reviews like this but I like flawed "ideological" design even less.

Velodyne Servo FSR-18

Velodyne Acoustics, Inc., 1070 Commercial Street, Suite 101, San Jose, CA 95112. Voice: (408) 436-7270. Fax: (408) 436-7276. E-mail: velodyne@earthlink.net. Web: www.velodyne.com. Servo FSR-18 servo-controlled powered subwoofer system with remote control, $2399.00 each. Tested sample on loan from manufacturer.

Our readers will recall that I reviewed the 15-inch Velodyne Servo F-1500R (very enthusiastically) in Issue No. 22. What we have here is not—repeat, not—the 18-inch version of the same design. That would have been the F-1800R, followed by the improved F-1800RII, both of which are superseded models at this point. I skipped both generations and ended up with the FSR-18, which is the latest and greatest Velodyne, David Hall's "statement" subwoofer, the flagship of a new line with redesigned drivers and class D amplification. (Even newer is the HGS-18, the same unit in black Formica, coming soon.)

I might as well say it right up front. This is far and away the most impressive low-frequency transducer in my experience as a reviewer. I can't imagine anything significantly better in a package of this size (footprint 21 by 18 inches, height 2 feet). What are the defining dimensions of subwoofer performance? Low-frequency extension, SPL capability, distortion. On the first two counts the FSR-18 is a top contender, on the last the undisputed champion. That makes it, all in all, the best there is. (At least in my book, until a manufacturer sends me something better.)

Outwardly, the FSR-18 is not very different from the 15-inch Velodynes, just slightly larger, with the same fairly austere styling. What is new is the 1250-watt (into 4Ω.) class D amplifier inside, but even that is being gradually phased into the rest of the line because it is compact and fits all models. (See the HGS-10 review below.) The class D approach is perfect for powered subwoofers because it is space-efficient, cost-effective, and without design tradeoffs when used only for low-frequency muscle. The controls and jacks on the back panel include the following: main power switch, subwoofer volume, lowpass crossover (120 Hz to 40 Hz), phase (0° or 180°), subsonic filter (15 Hz or 35 Hz), crossover in/out, auto turn-on switch (signal sensing on/off), external remote, line-level and balanced inputs, line-level output, speaker-level inputs and outputs, and highpass crossover (80 Hz or 100 Hz, 6 dB per octave) for both line-level and speaker-level outputs. Thus almost any system configuration is easily accommodated.

The main feature is of course the accelerometer-based servo system that controls the 18-inch driver (which, by the way, has tandem 3-inch voice coils in push-pull and a 14-pound magnet). The servo reduces the distortion to unprecedently low levels. I took an FFT spectrum of a 20 Hz tone right off the cone, with the 1-meter SPL set to 100 dB (out on the open floor—worst case). The 2nd harmonic was at -43 dB (0.71%), the 3rd harmonic at -46.5 dB (0.47%), the 4th harmonic at -52 dB (0.25%), the rest negligible. That's almost amplifier-like. I also took a THD sweep from 100 Hz down to 20 Hz, at a 1-meter SPL of 106 dB as normalized to 40 Hz. The THD fluctuated between 0.17% and 0.6%. Is that low enough for you? As for frequency response, my nearfield small-signal sweep located the f_3 (-3 dB frequency) at approximately 18 Hz, with the subsonic filter set to 15 Hz. (The filter cannot be switched out, only toggled between 15 Hz and 35 Hz.) Above the f_3, the deviation from dead flat is ±0.6 dB. With the crossover switched out, the upper "corner" of the frequency response is around 150 Hz. When pulsed, the FSR-18 shows a time-domain response corresponding to Q 0.7, which is correct damping for a sealed-box woofer.

How does the FSR-18 sound? Exactly as the above numbers would indicate. A subwoofer is totally deterministic. There are no mysteries, no surprises at the lowest audio frequencies. So, obviously: the amount of air moved is tremendous. The impact is tremendous. The resolution is tremendous. Bass drum thwacks are gut-wrenching. Organ pedal notes are seismic. Plucked string basses are ultradefined. More so than with less good numbers? That becomes an argument about the thresholds of human hearing, not about subwoofers. For $2.4K I'll go with the best numbers and leave the good-numbers-mean-bad-sound cultists to their own devices.

Velodyne Servo HGS-10

Velodyne Acoustics, Inc., 1070 Commercial Street, Suite 101, San Jose, CA 95112. Voice: (408) 436-7270. Fax: (408) 436-7276. E-mail: velodyne@earthlink.net. Web: www.velodyne.com. Servo HGS-10 high-gain servo-controlled powered subwoofer system, $1599.00 each. Tested sample on loan from manufacturer.

But there is neither East nor West, Border, nor Breed, nor Birth,/When two strong men stand face to face, tho' they come from the ends of earth!—KIPLING

The San Francisco and Seattle exurbs aren't exactly "the ends of earth," but in this case David Hall from San Jose and Bob Carver from Snohomish are definitely two strong men of audio, standing face to face in an engineering confrontation. What's it all about? Getting full-size subwoofer performance out of an 11-inch cube. Each of the two strong men believes his way is better.

Let's go back to January 1996. The Sunfire "True Subwoofer" has just been announced and a very early version of it is being shown at the CES in Las Vegas. I happen to run into David Hall and mention this new development to him. He already knows about it and is not impressed. He says something to the effect that he could easily do the same thing, only better, but he is not sure there's any point in doing it.

Fast-forward to fall 1996. The Sunfire subwoofer has been finalized and is in production; the dealers are selling it; the reviewers have their samples and are raving; the product is already a marketing success and getting hotter every month. I get my samples too and run my tests. In the spring of 1997 I publish a review in Issue No. 24, calling the design a tour de force and venturing the opinion that nobody but Bob Carver could have come up with it but at the same time quibbling about a few (non-major) flaws. Dr. Rich exceeds my own praise in his sidebar analysis of Bob's technology. And somewhere in the same time frame I recall David Hall muttering under his breath that maybe he will come out with an undersized subwoofer even though he doesn't have his heart in it, but this time it will be done right. (I wasn't surprised. How could the man who is Mr. Subwoofer USA ignore the rumored Sunfire sales figures?)

Well, I have an early sample of the mini-Velodyne and I have to eat my words. Bob Carver is not the only audio engineer who could have designed such a subwoofer. David Hall is another. Note that I say "could have" because David didn't—not until Bob demonstrated (1) that it was possible and (2) that it would sell. Bob had the original insight; David had the instant grasp of the principle. That principle—humongous magnet and humongous voltage swings to overcome the resulting back EMF—seems like elementary physics now that both Bob and David have demonstrated the viability of the design, but the approach was counterintuitive before its actual implementation and definitely beyond the ken of hack engineers.

Not that the Sunfire and Velodyne minisubwoofers are exactly alike. Actually, they are quite different and represent different priorities. (Bob Carver believes that the Velodyne is a total knockoff of his design, but I am not going to get caught in the middle of that controversy.) Let us compare.

Both units are 11-inch cubes. The Velodyne Servo HGS-10 is a sealed system with a single forward-firing 10-inch driver. It looks pretty much like a conventional subwoofer, only much smaller, with a grille cloth and a rather handsome black Formica finish. The Sunfire has a 10-inch passive radiator in addition to the 10-inch active driver and cultivates a no-grille, matte-finish, industrial-strength look. Both the Velodyne and the Sunfire drivers are extraordinarily heavy-duty, long-excursion units; they do not look exactly alike but are probably very similar in design. The Velodyne subwoofer uses an accelerometer for servo control through motional feedback; the Sunfire was also planned to operate that way, but after various problems with prototypes Bob abandoned the idea, and in the eventual production version only the circuit-board traces remained of the motional feedback system. The amplifier that powers the subwoofer is also of a different design in the Sunfire and the Velodyne; the former uses

Bob Carver's ingenious proprietary amplifier circuit with the tracking/switching power supply, the latter successfully employs a somewhat more conventional class D switching amplifier. (Bob's circuit has no particular advantage in a bandlimited low-frequency application; its virtues become important in a full-range power amp.)

The priorities reflected by these engineering differences are fairly obvious: the Velodyne goes in for very low distortion at all output levels, whereas theSunfire tries to squeeze the last bottom Hz and last dB of SPL out of that little box. At the risk of being a little too pat, one could say that the Velodyne is targeted at music, the Sunfire at home theater. In the end there is the inevitable convergence toward the performance limits set by the laws of physics—neither unit can move as much air as a big subwoofer with an 18-inch driver. What is remarkable is how much air they *can* move.

The features and controls of the Servo HGS-10 are pretty much the same as those of all current models in the Velodyne line. You can set the lowpass filter from 120 Hz down to 40 Hz; you can reverse the phase; you can bypass the crossover altogether (in favor of a separate crossover); you can choose between 15 Hz and 35 Hz "subsonic" filtering; you can use the built-in, nominally 80 Hz first-order highpass filter for your satellite speakers (strictly a minimal solution); and you can set the subwoofer for automatic turn-on in the presence of a signal. The variable lowpass filter is similar to that of the Sunfire in that it is rather low-Q in the transition band and steep only when the response is well into the stopband. The cleverly shrunk class D amplifier is the same as in the flagship FSR-18; it is said to be capable of 1250 watts continuous output into a 4Ω resistive load. (I did not surgically remove the amplifier from the enclosure to measure it separately because the only thing that matters here is the total system performance.)

My measurements revealed substantial differences between the Velodyne and the Sunfire. The latter definitely goes lower; the small-signal response of the Velodyne is -5.5 dB at 18 Hz, that of the Sunfire -1 dB at the same frequency. The -1 dB point on the Velodyne's response curve is at 30 Hz. On the other hand, because of the Sunfire's steeply plummeting low-frequency rolloff, the more conventionally profiled Velodyne actually has better small-signal response at 15 Hz (about -9 dB).

When it comes to distortion, the Velodyne is in an entirely different league, thanks to the motional feedback. The manufacturer's literature makes this claim: "At typical listening levels, the HGS produces less than 1% harmonic distortion with input signals extending to 20 Hz. At higher levels the distortion barely exceeds 1%..." I measured the HGS-10 under the most unforgiving conditions, radiating into a half sphere (corner placement would have made that an eighth sphere), with the crossover switched out and the measurement filter set at 22 kHz. I am sure the manufacturer's testing setup is kinder to the product. Down to 30 Hz, at 1-meter SPLs up to 97 dB as normalized to 40 Hz, I found the THD + N to be indeed under 1%. Sweeping down to 20 Hz and raising the SPL to 102 dB I was able to push the Velodyne into the 1-to-2.5% distortion band at the bottom of my sweeps. I also took an FFT of a 25 Hz tone at a 1-meter SPL of 98 dB and read 2nd/3rd/4th/5th harmonics of 1.4/1.1/0.16/0.45%, under the same brutal conditions. That's still well under 2% rms. I could have taken more distortion readings but I stopped once I was satisfied that the HGS-10 fits perfectly into the Velodyne Servo family, being almost as low in distortion as its big brothers. Quite a feat, knockoff or no.

There remains the question of maximum acoustic output, probably more important for car crashes, train derailments, and aircraft explosions than music. Louis D. Fielder of Dolby Laboratories, a highly respected psychoacoustics researcher with no speaker-marketing agenda (also a former president of the AES), has suggested 3% second harmonic, 1% third harmonic, and 0.3-0.1% higher harmonic distortion as the acceptable limits for accurate subwoofer performance. Under those limits the Velodyne is clearly capable of higher output in the 20 to 40 Hz octave than the Sunfire; at around 50 Hz the Sunfire catches up; without distortion limits the Sunfire with its passive radiator clearly wins over its full range by a few dB. To me the Arnold Schwarzenegger cinematic meganoise effects are a relatively minor issue. The fact is that within the typical dynamic range of modern recorded material both little subwoofers do an amazing job, and neither of them—let's face it—is quite as authoritative as a good 18-incher. (David Hall, who also sells 18-inchers, is more likely to share that perception than Bob Carver, who doesn't.)

"Comparisons are odorous," says Dogberry in *Much Ado,* and I'll end the comparison right here. For audiophiles with little room for equipment and big bass appetites, this is a win-win situation.

Waveform Mach 17 (followup)

Waveform, RR #4, Brighton, Ont., Canada K0K 1H0. Voice: (800) 219-8808. Fax: (800) 219-8810. E-mail: jotvos@waveform.ca. Web: www.waveform.ca. Waveform Mach 17 floor-standing 3-way loudspeaker system with electronic crossover, $6995.00 the pair (direct from Waveform). Tested samples on loan from manufacturer.

There have been some minor changes in this superb design, none of them important enough to necessitate a new model number or make a clearly audible difference.

The changes all have to do with ease of production and consistency from unit to unit. The "egg" is no longer made of fiberboard laminations shaped by lathe turning but is a massive one-piece aluminum casting, which is even deader acoustically and closer to optimal geometry.

The midrange driver is the same Audax unit with a special paper cone replacing the former plastic TPX diaphragm, which has been discontinued on account of manufacturing difficulties. The midrange contour control has been modified to complement the very slightly different response of the new driver. There is a barely measurable smoothing of the midrange as a result. A few other evolutionary changes, too small to dwell on, have also been made. Retrofitting is impractical and therefore not available, but replacement parts for the original production version remain in stock.

I have to reiterate that this is the speaker system with the lowest coloration and generally most livable-with sound known to me. Indeed, it has affected my critical perspective to the point where I must constantly remind myself to judge other speakers on a per-dollar basis because they all sound deficient, slightly or considerably, next to the Mach 17. And when it comes to the per-dollar criterion as applied to more expensive loudspeakers, my reaction turns to total puzzlement. Why would anyone spend more money for less accurate sound? Is today's high-end consumer still so naive as to believe that quality is directly proportional to price? Wake up, doctor, and smell the hype. (You too, counselor.)

In my original review I brought up the wild idea of adding a subwoofer (or a pair of subwoofers) to the Mach 17. I take it back. Don't do it. Further experimentation and in-room warble-tone measurements have convinced me that I was merely tipping up the bottommost frequencies for an impressive but unnatural effect. I was not extracting additional information, nor did I improve the upper bass and lower midrange as produced by the two 12-inch drivers. In my main listening room, which measures 22 by 20 by 9 feet, it is possible to position the speakers for flat response down to 20 Hz. That's as good as it gets. It sounds natural. The Mach 17 is a complete loudspeaker.

Our new contributor Glenn Strauss, who owns a pair of Mach 17's, disagrees with me regarding the subwoofer issue. His observation is that the two 12-inch drivers sound slightly cleaner when used only above 80 Hz in conjunction with an electronically crossed-over subwoofer and that the bass is deeper and flatter that way. I now think that's techno-overkill but of course I can only report the results in *my* room, with *my* test material. Other than that, Glenn fully agrees with my assessment of the Mach 17. It is a benchmark loudspeaker in terms of both sound and value.

Recommendations

I have serious misgivings about this kind of quickie wrap-up because it tempts you to peek ahead—didn't you?—before reading the full reviews with their all-important qualifications and reservations, thus inviting a jump to simplistic conclusions. On the other hand, this publication prides itself on taking unequivocal and unhedged positions on the subject of audio, so here goes...

Best full-range loudspeaker system we have tested: **Waveform Mach 17.**

Best minimonitor we have tested: **Joseph Audio RM7si Signature.**

Best powered subwoofer we have tested: **Velodyne Servo FSR-18.**

Best performance/price tradeoff in a *fully powered* full-range loudspeaker system we have tested: the combination of the **Paradigm Reference Servo-15a** with a pair of **Paradigm Reference Active/20s** as satellites.

Please note, however, that in the reviews above there are additional recommendations with more specific reference to available room, price, and individual requirements.

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...and another kind of transducer:

Sennheiser HD 600

Sennheiser Electronic Corporation, One Enterprise Drive, P. O. Box 987, Old Lyme, CT 06371. Voice: (860) 434-9190. Fax: (860) 434-1759. E-mail: lit@sennheiserusa.com. Web: www.sennheiserusa.com. HD 600 open-air dynamic stereo headphones, $449.95. Tested sample on loan from manufacturer.

I do very little headphone listening and am not familiar with a wide range of models. I have used various Stax electrostatics from time to time, as well as a few other high-end phones. On the basis of that admittedly limited experience I currently rate the Sennheiser HD 600 as the state of the art. Its sound is incredibly transparent, lifelike, dynamic, and uncolored. I hear nothing I would want to change. The elliptical earpads are very comfortable.

To measure headphones accurately one needs a dummy head with microphone ears; there is no other way. To purchase such a costly single-purpose device is not an option for this journal, so I'll just have to plod along with the crude technique I have always used. I simply hold the calibrated microphone close to the diaphragm in various open and blocked positions. Surprisingly, even this questionable method yields amazingly flat response curves with the HD 600. Up to 2 kHz the midrange and lower treble are literally ruler flat; the 2 kHz to 20 kHz decade is a little rough but shows no tendency to roll off on top. I suspect the little peaks and dips are an artifact of my measurement technique. The low-frequency response is ruler flat down to 80 Hz with a smooth decline to -8 dB at 20 Hz,which is probably due to the open-air design.

The impedance of this marvel is the standard 300Ω

There's Life Yet in the Two-Channel Integrated Amplifier

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

An analytical look at an often neglected but in the long run highly viable product category.

Integrated amplifiers are hot, particularly in the high-end market. The de-emphasis on phono (and these integrateds really de-emphasize phono) is part of the explanation. The move to higher-efficiency loudspeakers that require less amplifier power is another reason. We have not had access to any of the high-end integrateds made in this country and we are a little late testing Japanese integrated amps, since good ones have been available from the majors since the mid-'80s. They have been harder to come by in the '90s in the U.S. since the Japanese product in this particular category does not do well here. You will find no stereo integrateds in this country from Kenwood, Pioneer, Sansui, and Technics. Nor do top-of-the-line units from Denon and Yamaha make it across the Pacific.

Design Trends

Transistor integrated amplifiers have been in development since the '60s. As in their more expensive preamp and power-amp counterparts, significant improvements in measured (and maybe sonic) performance occurred as transistors improved in speed and power handling. In addition, circuit innovations took place that improved distortion performance. Improvements in protection circuits allowed complex loads to be driven without distortion. Improvements in output-stage topologies eliminated crossover notch problems. Unfortunately, just as cost-effective and transparent designs were about to hatch, the designers appear to have lost their way. Under the belief that their circuits did not sound good enough (controlled listening tests were just beginning to gain recognition at this time), they developed novel but not necessarily better topologies. This was the era of feed-forward, bridge-based distortion correction circuits, current-dumping topologies, class H output stages, adaptive output-stage biasing, and removal of global feedback loops. Even conservative design houses like Pioneer jumped in with both feet. Others like Yamaha, always driven by an itch to innovate, appeared to have a new topology every year. Last year's major breakthrough was replaced with this year's *really* major breakthrough.

It all settled down in the mid-'80s with a return to the simpler, rational circuits of the late '70s. Only Technics has held out with its class AA bridge-based amplifiers that attempt to separate the job of imposing a given voltage on the speaker terminals from the job of supplying the current required by the load to stay at the imposed voltage. In addition to the complex bridge at the speaker terminals, the design requires two power amps per channel. (We intend to look at this in greater detail if we can get a review sample.)

Unfortunately, the urge to tinker with perfection has again bit the Japanese designers in their latest integrateds. All the amplifiers below have something tweaky about them, and each shows reduced performance compared to its predecessors. You have to look no further than the manufacturers' data sheets to see this. My hope is that this new wave of "my amp sounds better than your amp because my amp uses a new topology based on Pyramid Power" will be over quickly. That said, none of the integrateds below have been so butchered as to have become audibly colored, and nothing has been done that should negatively affect reliability. I could live with any of them very happily and, indeed, I now own the Yamaha AX-592. I wanted a remote-based integrated for a bedroom system. The Yamaha was the cheapest and the AX-592's master remote ran the wonderful TX-950 tuner that is also sitting in the system. Now that you know how I make my purchase decisions... well, you may find that the long technical reviews are of limited interest, but I certainly hope you still find them useful for other reasons.

A Word about Phono Stages

If you are into phono, forget what comes with almost any current receiver or integrated amp. None of them are very good. To get very low distortion, very low noise, and very flat frequency response, a two-stage topology is needed. The best example is the one used by Bryston, but you have to purchase one of their premium preamps or integrated amps to get it. Great stuff if you can afford it, but if you cannot, Rotel has the answer. It is the **Rotel RQ-970BX** phono equalizer. The topology and components are lifted from the Rotel RC-980BX preamp that we reviewed in Issue No. 19. It is a huge bargain at $199.90. Even audiophiles should fall in love with its overdesigned power supplies, separate voltage regulators for each channel, and the premium metal-film resistors. We techie types fawn over the discrete bipolar diff pair with current-mirror tail coupled to an AD744 op-amp that provides the first section of gain. After a passive filter that sets the upper rolloff of the RIAA equalization properly (unlike the cheap single-stage topologies), the second gain stage provides the lower pole and the zero with an NE5534. What more could you want? A review sample has been promised to us, but in view of our laggard publishing schedule I see no reason for you to wait.

Denon PMA-2000R

Denon Electronics, a division of Denon Corporation (USA), 222 New Road, Parsippany, NJ 07054. Voice: (201) 575-7810. Fax: (201) 808-1608. Web: www.denon.com. PMA-2000R stereo integrated amplifier with remote control, $1000.00. Tested sample on loan from manufacturer.

Designers of Japanese hi-fi must all hang out at the same bar after work because new design ideas often appear in many companies' lines at the same time. For the last couple of years, MOSFETs have been "in." Besides Denon, we have Sony and Pioneer (in models available only in Europe). This is the second time MOSFETs are in. In the mid-'70s Yamaha, Sony, and Hitachi (the latter used to sell audio in this country) made a big splash with them. This was about the time when tubes were enjoying an increase in popularity, and one assumes the logic for MOSFETs was that they are more like tubes. Since tubes are again very popular, it is time for MOSFETs again. I predict that in a couple of years Japan will be back to bipolar and the tube mania in general will be over.

Denon first approached MOSFET design in their S1-series products, which are all priced close to or well into five figures. These designs are actually very good. First Denon observed that the really fast MOSFETs have low breakdown voltages, so they used these fast devices in conjunction with a dynamic cascode device that keeps the V_{DS} of the output stage to about 5 V. The cascode transistors are bipolar. OK, we now have fast MOSFETs but they still have low g_m . Denon has a solution here, too. The gates of the MOSFET are driven by an op-amp. The negative input terminal is connected to the speaker terminal, and the positive input terminal is driven by the power amp's voltage-gain stage. The op-amp power supply comes off the same line that drives the dynamic cascode output devices (±5 V). The gain of the op-amp brings the g_m down, and since the op-amp is flying up and down on the dynamic cascode rails, it sees no common-mode signal. Clever stuff!

Now on the POA-S10 power amp, which is about a 30th of the price of the S1 equipment, the op-amp is eliminated for some unknown reason. This saves the cost of a cheap op-amp (they used a TL071) and a handful of resistors and capacitors. The good news is they did keep the bipolar-based dynamic cascode. Now we step down to the PMA-2000R integrated under review here. Guess what got removed? You got it—the dynamic cascode. So it's back to the standard high-voltage MOSFETs. Indeed, it is back to the future because they use only NMOS devices in a quasi-complementary configuration with bipolar predrive transistors.

The quasi-complementary configuration disappeared in the early '70s. Those who go back that long will recall problems with stability and significant differences in the upper and lower transfer function of the output stage. Of course, with MOSFETs we have big transfer-function differences even in a complementary output stage, since the g_m of PMOS devices is higher than of NMOS devices for a given device size. The Baxandall modification to the bipolar quasi-complementary stage that did much to improve its performance is not used in the PMA-2000R, nor is it clear that it could be applied because of the use of MOSFET output devices. As we will see from the performance data, the Denon circuit works reasonably well, but one is left to ask why bother.

The rest of the power amp is standard Denon. Three sets of differential pairs form the voltage gain. The input pair is JFET-based; the rest is bipolar. The first two stages have resistive loads; the last is a current mirror. All three stages have resistor tails. The middle stage has emitter degeneration to reduce its gain to 10. Three gain-stage amplifiers can improve linearity because feedback factors are increased, but stability can be a problem if the engineers do not design this complex circuit correctly. Denon engineers appear to understand it very well. You will note that no mention of high feedback factors is found in the Denon literature. The company used to proclaim loudly that it built amplifiers with no global feedback. That idea appears to have been disposed of (instead, Onkyo has been bitten by the bug—see below), but Denon does not want to make a big deal about their use of feedback.

Performance of the amplifier was very good at the lower frequencies. It was noise-dominated into both 4Ω and 8Ω. loads. Clipping occurs at 180 watts and 120 watts, respectively. Distortion at that point is at -90 dB. At 20 kHz into 4Ω there are signs of dynamic distortion starting at 250 mW. Minimum distortion is -71 dB. Into 8Ω, dynamic distortion starts at 600 mW and the minimum distortion is -75 dB. Guess what I think causes the distortion. It is interesting to note that the Denon POA-S10 power amp (discussed above) is specified at $^2/_7$ the distortion level of this integrated. This is also the case, with only a very slight difference, for the bipolar-based PMA-1315R integrated that the PMA-2000R replaces. By the way, the optical bias circuit used in the 1315R, which Denon proclaimed for many years as a major breakthrough, is not used in the 2000R and the S-series line. Such is the fate of many circuit innovations that come from the land of the rising sun. I should also note that a similar situation exists in the case of the Sony TA-F707ES integrated (which we may review in the future). It has almost an identical topology to their older F700ES (a great integrated), except that MOSFETs (complementary in this case) replace the bipolars in the output stage. Sony specs report the distortion in the 707 to be 3.7 times higher, even though the power output has been reduced by 35% (the supply rails are the same but MOSFETs drop a lot more of the supply rails than bipolars do). The clear conclusion is that making MOSFETs work well requires novel topologies, like that of the Denon S1 series or the Cordell prototype (see Issue No. 20, p. 22).

The PowerCube of the 2000R is nice, with no signs of the 4-transistor protection circuit causing problems. It also shows that Denon has produced a very good quasi-symmetrical output stage with no signs of stability problems. Reactive loads always draw more voltage than resistive loads. With resistive loads, dynamic power goes from 139 W (33.3 V) into 8Ω, to 230 W (30.3 V) into 4Ω., to 334 W (25.8 V) into 2Ω, to 399 W (20 V) into 1Ω The large power transformer, large bridge rectifier, and low on resistance of the power MOSFETS used are responsible for this good performance. Generous heat sinks keep the unit cool during continuous power tests. As an aside, this unit does not have a speaker selector.

The phono preamp is built around the NJN4558 and a discrete JFET differential pair with just a resistor tail. You remember the 4558; it is the same op-amp that caused the measured performance of the Denon DCD-2700 CD player to be so poor (see Issue No. 22). Somebody at Denon must think it "sounds good" because they use the much better NJM2068 like tap water in their cheaper receivers. Because of its limited output current, the BA4558 cannot drive the RIAA network past 4 V rms at 20 kHz. At that output 20 kHz distortion is -74 dB. At lower frequencies the distortion is noise-dominated until the 8 V rms clipping level. The RIAA equalization error is only ±0.17 dB in one channel and ±0.1 dB in the other, but there is a channel imbalance of 0.3 dB in the upper bass and lower midrange. If phono is important to you, an external phono amp such as the Rotel discussed above should be used. By the way, in the $5200 PMA-S1 integrated, Denon decided to be the last of the big-time spenders and use the still very cheap NJM082 op-amp.

There is not much to say about the preamp because with the Source Direct switch activated there is no active preamp, just the selector switch and the volume control. This follows the approach of the PMA-1315R. In the direct mode no capacitors are in the main signal path. The nonpolar capacitor in the feedback loop of the power amp is wired in a novel configuration, so that it does not see significant displacement currents. The only downside of removing the active line stage appears to be increased noise levels. We have seen this in other integrated amps that are similar in this respect. Channel separation is 78 dB up to 1 kHz and decreases to 63 dB at 20 kHz. When the cheap balance and tone controls are put in the circuit, a unity-gain noninverting buffer using an NJM2068 also goes into the signal path. Switching in these controls takes the close-to-ruler-flat frequency response (-0.2 dB at 20 kHz) and puts it on a roller coaster that fits only in a strip 0.6 dB wide. Channel separation becomes about 8 dB worse with the controls in the signal path.

Overall build quality is good but not excellent. RCA jacks are all gold. The faceplate is quite thick, and the rest of the sheet metal is also of a heavy gauge. All that metal (heat sinks and power transformer included) adds up to the unit's net weight of 44 pounds. The selector and tape-monitor functions are implemented with sealed relays. The motorized volume control is of good quality. On the other hand, the PC board is single-sided; carbon-film resistors are extensively used instead of metal-film ones; and a cheap ribbon cable drives the low-cost tape-selector switch. The control knobs have a metal faceplate and a plastic body, but the overall look-and-feel is that of high-end equipment.

I picked a lot of nits in the above review, but overall this is a lot of amp for the price Denon charges and can be safely recommended; however, if you have a Denon PMA-1315R you have no reason to upgrade. Now, Denon, could we please have the S1 topology at the price of the 2000R.

Marantz PM-68

Marantz America, Inc., 440 Medinah Road, Roselle, 1L 60172-2330. Voice: (630) 307-3100. Fax: (630) 307-2687. Web: www.marantzamerica.com. PM-68 integrated stereo amplifier with remote control, $500.00. Tested sample on loan from manufacturer.

So you thought this must be a good review, since the Marantz MA700 power amp got such a good review in this issue. Think again. Different designer, different outcome. The PM-68 designer appears to be something of a tweak. How else can you describe a designer who kills the tape monitor in the direct mode (one assumes to reduce the number of switch contacts in the signal path, but when you count them up they are the same). Any competent repair person can clip one resistor and put the tape monitor back in the circuit for the direct path, but what on earth was the designer thinking in the first place?

Note that tweaky does not mean cheap. Input switching is by relay, which is a big surprise at this price point. That includes tape monitor and tape copy. Also of interest is the active preamp designed with a JFET differential stage and a resistive load, as well as an NJM2114 op-amp (a version of the 5532 with improved performance specifications, including lower noise). The volume control is placed close to this circuit to reduce crosstalk at the control's high-impedance output. Frequency response was down 0.3 dB at 20 kHz and for some strange reason had a small rise starting at 30 Hz that resulted in 10 Hz being up 0.3 dB.

Switching out Source Direct gets you tone controls and balance. The tone controls are complex second-order active types, each using one NJM2068 dual op-amp. Another NJM2068 buffers the tone controls' output. Channel separation in direct mode was 85 dB at low frequencies and 65 dB at 20 kHz. Switching in the all the controls changed nothing, indicating the crosstalk is occurring outside the signal paths of the control circuits. The separation numbers are excellent and should make some manufacturers of four- and five-figure audio electronics blush.

Let's look next at the phono section. It is the common JFET differential pair with resistive load. The op-amp that follows is the NJM2068. Distortion performance is typical of the breed. It is noise-dominated to the clipping point, which is at 7 to 9 V rms output (depending on frequency), where the distortion reaches a minimum of -82 dB. The earliest clipping is at 20 kHz as the op-amp runs out of steam driving the RIAA equalization network. The frequency response of the stage is down 0.9 dB at 30 Hz and 2.1 dB at 20 Hz. This is because only a 47µF deblocking capacitor is in the feedback path. I think this is some kind of tweak, not a cost saving, since we have not seen cost saving elsewhere. If you ignore the problem at the low end, the right-channel equalization error fits in a strip 0.4 dB wide, but the left channel needs a full dB. Channel matching, worst case, is 0.7 dB. No moving-coil option is available. That eliminates a usually cheap switch in a very sensitive low-level signal path, so I do not miss this feature.

The power amp shows a bit of tweaking around, and one wishes they had just scaled down the MA700 topology. The differential pair is a folded cascode instead of the telescoped cascode used in the MA700. The folded cascode adds a high-frequency pole. It is often used in circuits that work on very small (below 3 V) power supplies, but with 115 V to play with here one can only assume that somebody thinks it sounds good even if it measures worse. The second stage of the amplifier is also strange. Instead of the differential pair of the MA700, we get current-mirror interstage coupling. The loads of the first stage are a diode and a resistor that go to the negative supply rails. One side of the load is coupled to an *npn* transistor that forms half of the push-pull second gain stage. This transistor has an emitter degeneration resistor connected to the negative supply rail. This forms a current mirror. The current gain is set to 8 by the ratio of the degeneration resistor to the first-stage load resistor. The other half of the first-stage load drives a unity-gain current mirror, which has a diode-and-resistor load connected to the positive supply rail. This phase-inverting stage is then connected to the base of the *pnp* device in the second gain stage. The *pnp* is also resistively degenerated and like the *npn* side forms a current mirror with the previous (in this case the phase-inverting) stage. The current mirror gain is again 8. I suspect our tweaky designer views this whole circuit not as two gain stages and a phase inverter but instead as one gain block with current amplification stages. He must think that the one voltage-gain stage idea must sound better than two or three. The output stages are normal. Two stages of emitter follower buffers drive the dual-output devices. It is nice to see two buffer stages, since some amplifiers in this price class use only one. The protection circuit is the same as in the MA700. It is based on the Philips TA7317 custom IC.

The output bias setting circuit that is in series with the second-stage amplifiers is of interest because it allows two different bias settings in the PM-78 version of this amplifier available only in Europe. The alternate bias setting biases the amplifier into class A. When this switch is activated, the power-supply voltage to the power amp is significantly reduced.

Distortion curves are strange, although they are not that bad—but the MA700 is better. Into 8Ω the distortion curves show no dynamic distortion, but distortion becomes larger than noise at all frequencies starting at 20 watts, where it is down to -82 dB. It climbs slowly to -77 dB at 110 watts, where the amplifier begins to clip. Into 4Ω. the 20 Hz and 1 kHz distortion curves are similar, with a minimum of -80 dB at 40 watts and a -70 dB level at the clipping point of 160 watts. The 20 kHz distortion curve, on the other hand, does not show any flattening and looks noise-dominated. It reaches a minimum of -83 dB at 100 watts and then softly clips. At -70 dB distortion the amp puts out 180 watts at 20 kHz.

The PM-68 PowerCube shows that the protection circuit is overly sensitive. It is also responsive to the presence of reactive loads at 2Ω and below. I do not understand this since both the PM-68 and MA700 circuits use the Philips TA7317. Into 8Ω. the dynamic output is 149 watts when the load is purely resistive, with the voltage swing increasing slightly for reactive loads. The same is true into 4Ω, with the amp delivering 240 watts dynamic output into the resistive load. Into 2Ω the amp produces 300 watts with a resistive load and has slightly higher swings into a ±30° reactive load, but it will not drive reactive loads of ±60°, nor will it drive any load of 1Ω, resistive or reactive.

Construction quality is consistent with an integrated amplifier at this price point, as is the unit's overall look and feel. Everything considered, the Yamaha AX-592 wins by a nose. Its power amp puts out a little more power with a little less noise and distortion. Its phono section is more accurate, it has a separate tape-selector input switch, and it does not cut out the tape monitor in bypass mode.

On the other hand, the relay function selector of the PM-68 is significantly more reliable than the motorized unit in the AX-592. You really cannot go too far wrong with either unit.

Onkyo Integra A-9911

Onkyo USA Corporation, 200 Williams Drive, Ramsey, NJ 07446. Voice: (201) 825-7950. Fax: (201) 825-8150. E-mail: onkyo@onkyousa.com. Web: www.onkyo.co.jp. Integra A-9911 integrated stereo amplifier, $1119.95. Tested sample on loan from manufacturer.

Onkyo has always had a good conservative design team. Except for optical bias (used in the last generation of Onkyo integrated amps), Onkyo has not proclaimed a major breakthrough in design only to remove it two years later with a *really* major breakthrough that is gone in its turn. The A-9911 shows signs that this conservative approach is no longer being followed. The claim is that the power amplifier has no global feedback. What they really did was to split the power amplifier into a voltage-gain stage with a feedback loop and a current-gain stage with its own feedback loop. As will be seen from the amplifier's performance below, we can predict something new two years from now (as discussed above, the same thing was tried by Denon in the '80s and was forgotten in the '90s). The A-9911 does not have a remote control. The cheaper A-9711 does. One assumes the remote was eliminated from the A-9911 to allow a higher-quality volume control to be employed. Maybe they think an onboard microprocessor causes sonic problems. The other significant differences between the two units are that a smaller transformer, simpler protection circuit, and passive tone control stage are used in the A-9711.

The voltage gain section of the power amp in the A-9911 consists of a degenerated JFET differential pair with a three-transistor current source as the tail. A bipolar dynamic cascode is part of this differential pair. One load of the diff pair is a resistor, but the other is a Wilson current mirror that is terminated to the negative supply rail through a resistor. This circuit allows the complementary push-pull common-emitter second gain stage to be interfaced to the differential pair. A complementary pair of emitter followers with resistive loads buffers the output of the first voltage-gain stage from the input load of the second gain stage. The traditional V_{BE} multiplier is in series with the output of the second gain stage. What is not traditional is that feedback is taken off this second gain-stage output. We have now described the voltage amplifier.

The current amplifier is a fully complementary push-pull circuit. We shall describe a half circuit which consists of a common-emitter amplifier with resistive load followed by a source follower with a resistive load and finally a common-emitter amplifier that drives the speaker. Feedback is taken from the speaker terminal back to the first common-emitter amplifier at the amplifier's emitter. (We saw this in the Harman Kardon PA2400 power amp in Issue No. 21, but that one had the global feedback loop.) Two paralleled output devices are used. The series inductor usually placed between the speaker terminals and the output stage is missing. It is claimed the stage is so stable (because there is no feedback loop back to the input of the voltage stage) that the inductor is not needed. The use of inductor compensation between the first and second stage (also seen in earlier Onkyos) and the placement of the V_{BE} multiplier on the second-stage, and not the output-stage, heat sink (for improved thermal stability) are cited by Onkyo as important innovations over previous implementations of this topology.

While the concept looks very good on paper, the real-world distortion measurements were merely OK. Into 8Ω at the lower frequencies, distortion is -84 dB before clipping at 120 watts. The distortion is noise-dominated. Dynamic distortion with a 20 kHz signal starts at 3 watts. It reaches a minimum of -74 dB at 20 watts and then slowly climbs to -68 dB before clipping. It is a similar story into 4Ω.. Low-frequency distortion is -79 dB just before the clipping point (170 W). It is again noise-dominated. With 20 kHz the minimum distortion is -72 dB and at clipping it is -67 dB. This amplifier has no active preamp in the direct mode, and the power amp is set for a gain of 62. This may account for the relatively large noise that dominated our distortion measurements. The Denon PMA-2000R has a similar arrangement and also showed more noise than we are used to seeing, but it was better than the Onkyo. It is interesting to note that the Onkyo Integra A-809, which has a standard power amplifier topology (save the optical bias), is specified by Onkyo to have 7.5 times lower distortion than the A-9911.

A dc servo (a longtime Onkyo trademark) keeps dc out of the speaker. In direct mode the signal never sees a capacitor in the active signal path. A pair of bipolar transistors and an IC form the protection circuit, which opens a relay that disconnects the load if a fault occurs. There are actually two relays between the power amp and the speaker terminals. Thus, in addition to output protection, the relay pairs are used for speaker selection. A large power transformer and heat sinks (smaller than those of the Denon but still big) bring the amplifier's weight up to 44 pounds and help produce a good PowerCube, as we shall see below. The primary filter caps are 15,000 µF.

The PowerCube is nice down to 2Ω. The protection activated at 1Ω. and would allow no readings. Reactive loads always produced more voltage than resistive loads. With a resistive load, dynamic power goes from 130 W (32.25 V) into 8Ω, to 226 W (30 V) into 4Ω, and to 338 W (26 V) into 2Ω.

The preamp of the A-9911 has three modes. 'Power Amp' bypasses all but the volume control, 'Direct' includes the input selector switch, and 'Tone' switches in a discrete 10-transistor amplifier as well as the tone controls, balance control, and filter switches. The quality of the switches and controls in this signal path is not as good as that of the direct path. One really strange thing is that the volume control has four gangs (you do not get the extra pots in the cheaper A-9711). Two are used for the direct path and two are used for the tone path. Please do not ask why because I have no idea. Channel separation in the Power Amp mode is 86 dB at 200 Hz (maximum) and 54 dB at 20 kHz (minimum). In Tone mode things get only about 3 dB worse—a very impressive result.

The phono section is nothing to write home about. It is the standard discrete JFET front end with a resistive tail and an op-amp (a much better than average NJM5532). We see a cost reduction from the A-809, which had a dc servo and a current source tail. It takes a strip 0.7 dB wide to contain the RIAA equalization error. Things are flat from 30 Hz to 1 kHz, but then a 0.5 dB rise occurs, topping out at 12 kHz. Channel matching was very good at better than 0.1 dB. The NJM op-amp shows its stuff in the distortion test. The phono distortion is noise-dominated at all frequencies, with a minimum reading of-78 dB at clipping. The clipping occurs at 8 V rms output (measured at tape out).

The input selector is shaft-driven, but the switch does not look to be of very high quality. The recording selector is connected by a ribbon cable and it looks worse. The RCA jacks have gold grounds but their hot side does not. PC boards are single-sided. The volume control is very nice and metal oxide resistors are used in the power amp (smiley face goes here).

Overall, the A-9911 is a little pricey, especially considering it does not have a remote, but it can still be recommended. One is again forced to note, however, that the model's predecessor (A-809) looks to be the better unit. The cheaper A-9711 discussed above appears to give up little to the A-9911 and adds a remote control. It may actually be the better deal in the current Onkyo integrated lineup, but we have not tested it.

Yamaha AX-592

Yamaha Electronics Corporation, USA, 6660 Orangethorpe Avenue, Buena Park, CA 90620. Voice: (714) 522-9105. Fax: (714) 670-0108. Web: www.yamaha.com. AX-592 integrated stereo amplifier with remote control, $499.00. Tested sample on loan from manufacturer.

Shortly after the review of the AX-570 appeared in the last issue, this revised version of that design was released by Yamaha. From an external view it looks like a cosmetic update. Speaker connectors that have EEC approval are at the rear. There are two types of connectors. For Speakers A we get plastic push-in connectors on steroids that are not fun to use. Speakers B get banana jacks with spacing such that you can no longer connect double banana plugs. At the front we have a new sliding door that hides secondary controls and gives the unit an upmarket look. The remote is also revised. The rows of identically sized/colored/spaced buttons of the old control are replaced by a new control that is ergonomically friendly. This time we have three colors and five distinct button shapes, plus a novel pistol-grip housing. A gold star to Yamaha for the remote redesign.

I wish they had stopped with the cosmetics, but unfortunately they mucked with the internals of this unit and the result is a significant reduction in performance. First they removed the direct ground-sensing circuit from the power amp. This is a typical move for Yamaha, where last year's breakthrough is gone with the next model. Remember the ALA circuit, the Hyperbolic Conversion Amplification circuit, the X amplifier, the Linear Transfer Bias circuit, the Extended Rolloff Equalizer, the all-MOSFET power amplifier, Auto Class A, and the ever popular Zero Distortion Rule circuit?

The gain of the power amp has been changed from 28 dB to 36 dB because the preamp gain has been lowered, as we shall see below. The power amp is otherwise identical to that of the AX-570, except for a major cost-cutting move. The bias trim pot has been replaced with a two-position jumper. In the AX-570 the trim pot was adjusted to keep the bias current within 40% of the nominal value. The jumper allows for a 20 to 1 variation. The result was a significant worsening of distortion performance. It still meets the specs but performance was now average at best. The high dynamic distortion was a telltale sign that a bias trim was in order, but with this amp it was impossible to do. Distortion into 8Ω reaches a minimum of -95 dB with a 20 Hz input but is already 1 dB higher with 1 kHz. With signs of dynamic distortion at such a relatively low frequency, we expect and get much worse results at 20 kHz. There the distortion starts to exceed noise at an output of 50 mW! It gets as low as -84 dB at 30 watts, then slowly goes back up to -80 dB at the onset of clipping. The amp puts out 140 watts before it clips. Into a 4Ω. load clipping occurs at 210 watts, with distortion running 2 to 3 dB worse across the board. What is amazing is that despite the misbiasing, this power amplifier topology is still producing less distortion than any other amp in this issue save the Marantz MA700.

PowerCube results were more encouraging. Dynamic power into 8Ω was 36 V rms (163 W) with a resistive load and 37 V rms (171 W) with a highly reactive load. The good protection circuit of the AX-570 was retained and works well. Into 4Ω the voltage drops slightly to 32.6 V (266 W) with a resistive load and is again a little higher with reactive loads. Into a resistive 2Ω. load the voltage has dropped to 26.8 V rms (360 Watts) and again rises slightly higher into ±30° reactive loads. The $2\Omega/60^0$ results do show a drop in voltage as the protection circuit starts to activate. Into a $2\Omega/60^0$ inductive load, 20.5 V rms (210 W) is all the protection circuit would allow. Into 1Ω/0° the voltage has dropped to 19.6 V rms. The protection circuitry would allow only 11.5 V rms into a 1Ω/±30° load, and 8.2 V rms is all it would allow into a 1Ω./60° inductive load. Overall a very impressive PowerCube for a $499 integrated amplifier—it could make some four-figure high-end power amps blush. All of the foregoing power-amp measurement results apply only to the unit's high-voltage supply-rail setting. A new switch on the back of AX-592 allows you to lower the rail settings for use with low-impedance loads. I suspect this is another EEC safety move. Power drops to 70% of the values reported above, but the protection circuit allows this full power into 2Ω/±60° loads and 1Ω/±30° loads. Bring on those high-end loudspeakers with their killer impedances! And if you live in Europe and need more power, try the AX-892. It has different output devices, bigger heat sinks, a different transformer, and different main power-supply filter caps but is otherwise identical to the AX-592.We will not be testing it because it is not available in the U.S.

The preamp has undergone significantly more change than the power amp. A 6-transistor complementary discrete amplifier is connected directly to the CD inputs and provides a gain of 11 dB. Why anybody would want to do this is unclear, since this very simple little amplifier will swing 10 V peak if a CD player's maximum output is 2 V rms. If you have a CD player with a higher output, you could easily clip this amplifier, which lives on 14 V supply rails. A new switch called 'CD Direct' routes the signal from the output of this amp directly to the volume control and then from the volume control directly to the power amp. If you do not activate CD Direct, then you can select CD on the function selector. In this mode the signals are routed directly from the function selector to the volume control, and then comes the gain of the 10.4 dB line amplifier. (This time a simple NJM2068 op-amp with three electrolytic dc-blocking capacitors—in the input, output, and feedback loop. The discrete CD amplifier dispenses with the feedback cap.)

In the AX-570 the active amplifier was on the same board as the volume control, with virtually no interconnect wiring. The result was exceptional channel separation. Now the volume control and the amplifier are on different boards, with lots of wiring, including a trip up to that CD Direct switch. Guess what happened to the channel separation. It is no better than 85 dB and it decreases at 20 kHz to 40 dB. Switching to CD Direct does nothing to help things—as expected, since the high-impedance signal at the output of the volume control still goes through lots of wiring and switches. We did see the 0.4 dB level difference between normal and CD Direct modes, a direct result of the different gains of the two amplifiers. That is just the right number to make people think they are hearing a quality difference in CD Direct, even though it is just a gain change. Frequency response with the tone controls bypassed is flat to 1 kHz and then rolls off to -0.3 dB at 20 kHz. That rolloff may be a result of the higher gain setting in the power amp. Put the tone controls in and things get lumpier, with the response moving from -0.2 dB at 20 Hz up to 0 dB at 50 Hz, then down to almost -0.2 dB at 300Hz before going back to -0.1 dB at 1kHz.

The RIAA phono section is unchanged from the AX-570. The less good channel had an equalization error running from —0.2 dB at 200 Hz up to +0.3 dB at 20 kHz. Worst-case interchannel matching was 0.25 dB. No distortion above the noise level could be identified below clipping in this phono stage. Noise was about 10 dB lower in this stage, which uses a discrete bipolar front end, than in the FET-based stages used by the other integrated amps reviewed here. Of course, the FET-based stages do not need an input dc-blocking capacitor, but given the small signal size coming into the circuit this would not appear to be significant.

Nothing we found in our measurements should affect the audible performance of the Yamaha AX-592, but it is clearly no longer the remarkable performance bargain the AX-570 was. I called the AX-570 the Toyota Camry of amplifiers. The AX-592 is more like a Ford Taurus. It has a prettier face but worse performance when you road-test it. •

How to Be a Sophisticated Audiophile *(continued from page 17)*

some useful information on the subject in this issue, but with our sparse publishing schedule we are unlikely to have all the answers when you need them. Tread carefully on this ground.

FM, DACs, and Whatnot

One area of audio electronics where the differences are not only major but unrelated to price is FM. The engineering of tuners, and of the tuner sections of receivers, covers the gamut from el cheapo to very impressive—and not at all in the ascending order of price tags. The FM articles and 10 tuner reviews in Issues No. 23 and No. 24 should steer you in the right direction. What I find particularly amusing is that none of the passion that goes into audiophile debates about nonissues like, say, CDs versus vinyl is evident on the subject of FM reception, which is a genuine issue affecting most home music systems.

Speaking of CDs, I haven't run into a truly bad CD player for years. Of course, I haven't tested anything under $400. There are huge differences in the DAC chips as well as in the various op-amps in the signal path, but the players that measure only so-so also do an audibly impeccable job. Sorry, vinyl diehards, but in today's audio world the CD *rules.* (I haven't tested a sufficient number of DVD/CD players to make such a sweeping

(continued on page 72)

An Alphabet Soup of Electronics: AV, DVD, CD, FM, THX, AC-3, DTS (all A-OK)

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

More channels, more complexity, more stuff to buy, more money. The surprising thing is how good most of it is, despite the hype.

Editor's Note: *The electronic circuitry we are dealing with here falls into three basic categories: analog signal paths, A/D and D/A conversion, and digital signal processing (DSP). We are accustomed to testing the first two; the third cannot be tested under our usual protocols. DSP involves computer chips (CPU's) and algorithms; the latter vary from brand to brand and designer to designer even when the audio format (say, Dolby Digital) is the same; and the discipline of evaluating the end result is far from routine, not to mention time-consuming. Most reviewers do it unscientifically, by the seat of their pants. One exception is David Ranada, whose infrequent test reports in* Stereo Review *on DSP-intensive equipment are quite exhaustive and authoritative. We are refraining herefrom making any statements about the DSP sections of AV equipment that we cannot back up with laboratory test data. Our educated guess is that so far none of the equipment is capable of totally transparent DSP, in the sense that a stereo line stage, for example, can be totally transparent to analog signals. As the technology matures, we can expect a convergence toward transparent solutions. (David Rich's comments on the special problems of AV/DSP testing appear within the individual reviews.)*

AV Surround Receiver

Denon AVR-5600

(Reviewed by Peter Aczel and David Rich)

Denon Electronics, a division of Denon Corporation (USA), 222 New Road, Parsippany, NJ 07054. Voice: (973) 575-7810. Fax: (973) 808-1608. Web: www.denon.com. AVR-5600 AV surround receiver, $2800.00. Tested sample on loan from manufacturer.

The tacit statement made by Denon's flagship AV surround receiver is that you can have it all on one chassis and still enjoy the highest level of performance, such as you would obtain with a separate AV controller/processor and five outboard power-amplifier channels. The statement, if actually voiced and then interpreted literally, is disputable (how about, say, a Lexicon DC-1 with a Bryston 9B ST?), but the fact remains that the needs of even a rather demanding home-theater enthusiast are quite nicely served by this well-thought-out and THX-certified piece of equipment. The only thing missing is DTS, not exactly a disaster. Interestingly, a lot of flagship-type high-powered AV receivers appear to be priced at $2800 (B&K, Onkyo, etc.). So far, this is the only one to come our way. I understand that an improved/updated AVR-5700, this time with DTS and other goodies, is in the pipeline.

The AVR-5600 accommodates 11 program sources and every extant surround format except, as I said, DTS. The source and format are clearly displayed on the TV screen, not only on the front-panel readout. The volume level is indicated in glowing red numbers on the front panel. A very explicit AC-3 channel indicator on the front panel is a particularly nice feature. The remote control is highly versatile and easy to use. Hooking up an AV system is a cinch because of the well-integrated design.

The power amplifiers, as measured in the direct mode at 23-times gain (00 display), are low in distortion but could be lower. The THD + N curves are not entirely noise-dominated. Into 8Ω the 1 kHz curve bottoms out at -87 dB and 45 watts, the 20 Hz curve at -85 dB and 38 watts, the 20 kHz curve at -78 dB and 7 watts (dynamic distortion). The THD gently rises after these minima; maximum output just before clipping is approximately 160 watts at all frequencies. Into 4Ω the curves have almost the identical profile at proportionately higher output; clipping occurs in the vicinity of 280 watts. The small-signal frequency response of the power amps in the direct mode is -0.3 dB at 20 Hz and 30 kHz, a bit more rolled off than the best; with the controls in the signal path a saddle appears: +0.3 dB at 170 Hz, +0.1 dB at 5 kHz. Channel separation between the front left and right power amps at 1W/8Ω is 53 dB at 20 kHz, increases at 6 dB per octave to 88 dB at 90 Hz, and drops back to 81 dB at 20 Hz. That's very good for a crowded chassis. Switching the controls into the signal path has only a tiniest effect on these figures; that's even better.

The PowerCube of the power amps paints a fairly decent picture. All reactive loads draw a slightly higher voltage than the corresponding resistive loads, as they should. Dynamic power into 8Ω/4Ω/2Ω/1Ω. resistive loads is 212W/338W/398W/283W (41.2V/36.8V/28.2V/16.8V). Not the ultimate in current capability but far from bad.

The analog circuitry of the preamp section also yields decent measurements. At maximum gain (16.4 dB, 6.6 times) in direct mode, the CD-in/pre-out distortion curves bottom out at -81 dB, and the clipping point is 9 volts. At unity gain, the lowest distortion at all frequencies occurs at 1.2 volts and measures -84 dB. The MM phono-in/tape-out distortion also bottoms out at around -80 dB, regardless of frequency, with 6 volts maximum output before clipping. RIAA equalization error is ±0.2 dB.

On the other hand, the D/A converter measurements are a bit on the shabby side. Coax-in/pre-out frequency response is -0.6 dB at 20 Hz (otherwise flat). Full-scale distortion of the DAC is -85 dB in the midrange and wildly fluctuating above 3 kHz (don't even ask the numbers). With the digital input reduced to a less brutal -20 dB, the DAC still looks like a 15-bit job. Low-level linearity error is -3.2 dB (!) at the -90 dB level in one channel, -1.0 dB in the "better" channel. The "Rob Watts test" (FFT spectrum of a dithered 1 kHz tone at -60 dB) reveals more than trivial odd-order harmonics all the way up to 9 kHz.

—*Peter Aczel*

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This unit is essentially similar to the much cheaper AVR-3600. That one has no fan, a cheaper ADC, one less DSP (bass management becomes an analog thing), no tone controls for the center and the .1 channel. Going down to another level to the AVR-3200 brings us to a very different design that is clearly targeted at a different market segment.

The tuner front end of the AVR-5600 is mystery meat. The IF strip consists of one transistor and two filters. The limiter/demodulator and multiplex decoders are separate Sanyo chips—things could be a little worse but not much. If you want good FM from anything but the local rock station, expect to add on a separate tuner. As Denon's David Birch-Jones points out, that is actually the only way to go, since all the digital noise inside an AV receiver makes putting a high-performance tuner inside impossible. On the other hand, the high-end AV receiver from Onkyo we looked at in Issue No. 23 had a more complex, better performing tuner than this Denon.

Analog input selection gets the deluxe treatment. Every input is buffered by an op-amp to prevent latchup in the CMOS selector switch. This expensive approach reduces distortion in comparison with using a resistor to drive the CMOS switch directly. Good-quality NJM2068 op-amps are used in most of the signal paths, but even better-quality NJM5532's pop up in some of the L and R signal paths. The ADC is an oversampling Nyquist A/D converter (PCM 1760) from Burr-Brown. A separate DF1760 does the decimation. Denon is apparently allergic to delta-sigma ADCs and DACs. DSPs are the Zoran ZR38500 and Motorola DSP56004. That sounds impressive, but since I have no knowledge of how they are programmed, I have no way of telling if they do anything nasty to the signal. Remember, with a different software program the same DSPs might be in a smart telephone or a guided missile. To control things, the receiver has three microcontrollers (if you include the one in the remote). The AC-3/RF demodulator looks about as complex as the FM tuner. DACs are Burr-Brown PCM69's (lowest grade). This is the same DAC you get in the $999 AVR-3200 and it performs like a DAC from a $200 CD player. This cost-cutting move is a performance killer. NPC5841's drive the DACs. *I/V* conversion and reconstruction filtering is done by two op-amps.

Rear, center, and the .1 channel go to a digital volume control formed with two op-amp stages and a Sanyo digital potentiometer (two pots per channel). The left and right get better treatment. The output of the DAC reconstruction filter or the analog signal from the analog input selector (if you are in direct analog mode, in which case you had better have full-range speakers because the .1 output gets killed) enters a single-ended-to-differential converter. The digital volume control is implemented differentially. Essentially, this is two of the stages used for the center and rear channels working for one channel. After the digital volume control, differential-to-single-ended conversion is performed. The use of balanced volume controls reduces distortion from the monolithic digital potentiometers. Balanced signal paths will also allow you to become a card-carrying member of the high-end audio club. Relay switching of the cheap analog tone controls also lets you keep the card. The rear channels solve the tone-control defeat problem in a different way. No tone controls are in the signal path. I have no idea why. Note that with the tone controls defeated, there is not a single mechanical switch or pot in the signal path.

What's not from the high-end club is the power amp. A differential pair with resistive biasing and load drives a second diff pair with resistive biasing and an active load. This drives the one set of predrivers, and then it is on to the single output devices, which one assumes are some form of composite device in a single package, given the power this unit can sink. While this sounds about as simple as you can get, Denon knows how to make things even simpler in the AVR-3200. The AVR-5600 amplifier swings a lot of voltage but not much current into low-impedance loads. The current limiter is slightly touchy and may be responsible for the lower output numbers into the 1Ω. load. Distortion performance is consistent with the power amp design. The power amps live in a very crowded part of the neighborhood inside the receiver, and that may be responsible for some of the dynamic distortion. Denon knows how to do better, but you have to pay more. Their POA-8200/8300 power amp gives you five nice, if pricey, 120-watt channels that are better engineered than those of the AVR-5600 and have none of the tweaky aspects of Denon's two-channel stuff (no MOSFETs here).

*—**David Rich***

DVD Video Player

Denon DVD-3000

(Reviewed by Peter Aczel)

Denon Electronics, a division of Denon Corporation (USA), 222 New Road, Parsippany, NJ 07054. Voice: (973) 575-7810. Fax: (973) 808-1608. Web: www.denon.com. DVD-3000 DVD video player, $899.00. Tested sample on loan from manufacturer.

This is only the second DVD player to come into my life; the first was the Sony DVP-S7000 reviewed below. The Denon is "second generation" and very nice, but I see nothing on the screen when I play DVDs that would make me rank it above the first-generation Sony. The menus are perhaps a little less confusing but barely; the construction is definitely more flimsy for only $101 less; there is onboard 5.1 decoding/outputting which no one with an even moderately sophisticated home-theater front end is likely to use; there is component-video output but that's expected at this price. My quick take on this product is that that Denon just had to have such a unit in the line, so they OEM'd this one from Matsushita (a very respectable source) while getting ready to launch their "reference" DVD-5000 ($2500). The latter should be the moment of truth for Nippon Columbia's DVD playback technology; meanwhile I'll just report my measurements of the DVD-3000's CD performance. (Please see the Sony DVP-S7000 review below for a brief discussion of the limitations of our approach to AV testing.)

Frequency response is -0.2 dB at 9 Hz and 20 kHz. Below 150 Hz, THD + N at full scale is about 18 dB (!) in excess of the 16-bit theoretical ideal of -98.08 dB, strangely improving to 5 to 7 dB in the midband. It must be gain-related analog distortion, but I've never before seen a worst-case situation at the lowest frequencies rather than higher up. Reducing the digital input level drastically results in only 1 dB excess distortion at 1 kHz; however, I couldn't run the test at other frequencies because the Denon player refuses to recognize and play the 99-track test CD I use for that purpose. It says "no disc" when that particular CD is in the drawer—another first in my experience. One would suspect the laser tracking assembly, but the error correction of the player is just as good as that of the Sony DVP-S7000 according to the new Digital Recordings test disc. (Both are good; neither is perfect.) Gain linearity is error-free (±0.0 dB) down to the lowest levels, well below -100 dB, a Japanese near certainty these days. Quantization noise is -94.7 dB (good but not the best); dynamic range is 97 dB (close to the best). The FFT spectrum of a dithered 1 kHz at -70 dB (modified "Rob Watts test") shows a definite 7.5 kHz glitch, 15 dB high, rising out of a bin-by-bin noise floor of -127 dB. That's a tad short of squeaky-clean. The monotonicity pattern is slightly more irregular than the best I've seen but still OK.

Bottom line: if I had no other DVD player than the Denon DVD-3000 I would live with it quite happily, without significant complaints, but at this point in its model life I wouldn't go out and buy it. I would wait for the next generation.

Digital Surround Processor/Controller

Lexicon DC-1

(Reviewed by Peter Aczel)

Lexicon, Inc., a Harman International Company, 3 Oak Park, Bedford, MA 01730-1441. Voice: (781) 280-0300. Fax: (781) 280-0490. E-mail: info@lexicon.com. Web: www.lexicon.com. DC-1 digital controller (THX/Dolby Digital/DTS version), $4995.00. LDD-1 RF demodulator for laser disc AC-3 output, $699.00. Tested samples on loan from manufacturer.

There are DC-l's and there are DC-l's. It's like a Detroit car; you load it up with options on the salesman's order sheet. The stripped-down "base" version costs only $1995.00 and lacks THX, Dolby Digital (AC-3), and DTS facilities. The over-the-top version, which is the subject of this far from exhaustive review, costs almost three times as much (if you include the outboard RF demodulator) and looks exactly the same. Indeed, there is even a vague outward resemblance to the old CP-3 PLUS, reviewed in Issue No. 23. That one already taxed my reportorial resources with its profusion of multichannel audio processing features, but a loaded DC-1 beats it handily.

A complete technical analysis of the DC-1 a la David Rich would require detailed circuit schematics and DSP programming information, neither of which we have, and take up too many pages of this issue, so I was rather relieved that he did not undertake such a project. (It's easier not to unleash David than to rein him in—if I may mix my canine and equine metaphors. Besides, he keeps talking about a floating-point to fixed-point conversion screwup in the design, which is the scuttlebutt of DSP engineers he knows and which I don't want to touch with a ten-foot pole until I have a better handle on it and can ascertain its existence.)

Suffice it to say that there is no multichannel control unit known to me that will implement more surround-sound formats than this version of the Lexicon DC-1. I counted 24 (but that includes various stereo and mono formats as well). My criticism of the CP-3^{PLUS} must be revised for its successor; the DC-1 will do very nicely as the sole control center of just about any home entertainment system, whether the emphasis is on music or video. There are eight analog inputs, five of them with video (including three S-video), and four digital inputs (two coax, two optical). There are outputs for eight audio channels: front left, front right, center, subwoofer, side left, side right, rear left, rear right (Lexicon believes in, and promotes, 7.1 surround sound). It's much more convenient to have the TV on at all times in order to keep track of all this via the on-screen displays; the front-panel display is very nice but limited to two lines, wheras the TV screen shows the huge menus in their entirety. (Hey, the life of an advanced AV enthusiast is not an easy one.) Unfortunately, the ergonomically well-designed remote control doesn't seem to communicate reliably with the menus in a room with fluorescent lighting—be forewarned.

As you can imagine, the electronics that make all this versatility possible are extremely complex; maybe only David Griesinger understands every little detail of it; I certainly don't. All I know is that entirely new chips may appear in the DC-1 at any time during its production life without a change in the basic model designation or even the addition of a model suffix—Lexicon believes in making continuous small improvements undisturbed by every geeky must-have-the-latest customer. Thus the performance of the DC-1 will have to speak for itself. (I also refer you to David Rich's explanation of why the digital surround algorithms cannot be critiqued by us; see his review of the Sony STR-DA80ES receiver below.)

That brings me, willy-nilly, to the measurements. In my CP-3^{PLUS} review I observed that there appears to be a -80 dB "stone wall" in the line-level THD + N performance of all AV surround electronics. The DC-1 does not break down that wall; with the volume set at 0 dB, the front left channel reads -80 dB at an output of just over 2 volts (except with a 20 kHz input, which reveals a few dB of dynamic distortion, not enough to worry about). Frequency response is flat within ±0.1 dB from 20 Hz to 20 kHz (even with who knows what complexities in the signal path); front left/right channel separation is excellent: 85 dB at 20 kHz, increasing to 104 dB at 20 Hz, in the less good channel. The noise floor of the front left channel with shorted input is in the 100 nV to 1 µV range from 20 Hz to 60 kHz, except for a 60 Hz power-supply leak of 4.5 µV.

The D/A measurements at 44.1/16 indicate DAC quality almost on the level of the best CD players and outboard D/A processors. At full-scale output the THD + N reads -91.8 dB across most of the audio spectrum, the 6.28 dB excess distortion above the theoretical ideal of -98.08 dB being mostly gain-related analog distortion. With the digital input only slightly reduced, the excess distortion shrinks to 2.8 dB. The "Rob Watts test" (FFT spectrum of a dithered 1 kHz tone at -60 dB) shows absolutely no harmonic blips from a bin-by-bin noise floor of-125 dB. Gain linearity is outstanding all the way down to the -100 dB level, with less than 0.2 dB error even below -90 dB.

The most relevant question one can ask about the Lexicon DC-1 in this all-out version ($5694.00 with the outboard RF demodulator) is undoubtedly this: Is it just another high-end fantasy product for the moneyed audio/videophile, or is it a tool capable of functions that are not performable at lower cost? I tend to side with the latter point of view. The 7.1 surround capability is just one reason (yes, it sounds better than 5.1). There is the Logic 7 encoding/decoding capability, which makes all formats—mono, 2-channel stereo, Dolby Pro Logic, 5.1 Dolby Digital, etc.—forward and backward compatible without loss of balance, bandwidth, dynamics, and directional cues. There is the uniquely comprehensive bass management capability (a weakness in nearly all other AV equipment). I could go on but I promised from the start that I wouldn't.

Bottom line: the Lexicon DC-1 is clearly in the audio/video vanguard of the digital era; I only wish the price would come down as it has in equally complex PCs.

FM Tuner

Magnum Dynalab FT-101A

(Reviewed by David Rich)

Magnum Dynalab Ltd., 8 Strathearn Avenue, Unit 9, Brampton, Ont., Canada L6T4L9. Voice: (905) 791-5888 or (800) 551-4130. Fax: (905) 791-5583. E-mail: magdyn@myna.com. Web: www.magnumdynalab.com. FT-101A analog FM tuner, $875.00. Tested sample on loan from manufacturer.

After our very satisfying experience with the Magnum Dynalab 205 "Signal Sleuth" antenna amplifier, Richard Modafferi and I were expecting good things from their complete tuner. Alas, it turns out not to be so. This is a straightforward minimal design using LSI integrated circuits from National and Philips and a single-transistor RF stage. The front end has only 4 tuned elements. The input network being singly tuned, it came as no surprise that the RF overloaded with the 1 V signal on 92.1 MHz and even with the 0.25-volt signal on 105.7 MHz. These stations ruined reception through the bottom third and top third of the frequency range, respectively. Adding in the Signal Sleuth does not make things better, since it too overloads on these big signals.

Only three ceramic filters are visible in the IF section, all three in sockets—perhaps to allow easy "kitting" of individual filters for best performance. In the narrow mode selectivity was not good enough for serious DX reception. Given the poor RF performance, we could not do any selectivity use tests on the outdoor antenna. Moving to an indoor ¾-wave antenna revealed that DX reception of difficult adjacent-channel signals was still not possible because of the insufficient selectivity. Adding in the Signal Sleuth did not help things much. The killer test signal on 91.3 MHz was quieter but still had crosstalk from the local on 91.5 MHz. We did get the distant station in Canada on 105.7 MHz using the combo. (See Issue No. 23, page 58 for more information on the use of these stations for testing.) In a crazy experiment, having no schematics and only guessing what circuitry surrounds the FT-l0lA's IF filters, Richard tried plugging a Mcintosh MR-78 supernarrow filter into each of the three sockets holding the original ceramic filters of the FT-101 A. It worked on one! Who says you need good impedance matching in RF circuits? The killer signal on 91.3 MHz was received clearly.

Not surprisingly, measured performance was nothing to write home about, given the unit's parts. THD in the wide mode was poor at -51 dB, and 10 kHz IM distortion was -64 dB. In the narrow mode things did not get much worse (things are of course not so good to begin with, and the narrow mode is by no means narrow). THD was -49 dB and 10 kHz IM was -60 dB. It will come as no surprise to those of you who followed my tuner reviews that the FT-101A uses a quadrature detector. In the wide mode, channel separation is 45 dB at 1 kHz, dropping to 35 dB at 10 kHz. In narrow mode, the 1kHz and 10 kHz channel separation is 35 dB. The folks at Magnum Dynalab have asked us to make you aware that we adjust the signal generator carrier frequency for minimum distortion. They point out that in frequency-synthesized tuners this is not fair, since there is no way to make small changes in center frequency at the tuner. No way except with their tuner, since it is not frequency-synthesized. The tuning knob is in reality a variable power supply that drives the varactor directly. The frequency display comes from a frequency counter. Of course, this system is subject to drift, and modern frequency-synthesized tuners have spacing as small as 5 kHz, making the problem almost moot. Then again Yamaha solves the problem nicely by using a combination of PLL synthesis and an advanced form of AFC.

To end on a happier note, I should report some new use-test data Richard Modafferi obtained on the **Magnum Dynalab 205 Signal Sleuth.** Richard was not involved in the original test of the 205, but we sent it to him with the FT-101A on the assumption that the best results might occur when the two units were paired. After the FT-101A turned out to be no bargain, Richard tried the 205 with his Mcintosh MR-78 reference tuner and an indoor vertical antenna. He reports he was able to receive the killer test signal on 91.3MHz. He reports "the performance of the 205/indoor-vertical combination was pretty amazing, as it was used in a ground-level room inside an aluminum-sided house. Use of the 205 to enhance reception on an indoor antenna can be highly recommend." OK, Magnum Dynalab, we know you can perform RF magic as shown by the 205, and we know that substitution of better IF filters significantly improves the performance of the FT-101 A. It is time for you to get to work and produce a tuner of high quality.

Dolby Digital AV Preamp/Tuner

Marantz AV550

(Reviewed by Peter Aczel)

Marantz America, Inc., 440 Medinah Road, Roselle, IL 60172-2330. Voice: (630) 307-3100. Fax: (630) 307-2687. Web: www.marantzamerica.com. Model AV550 AV preamplifier/tuner, $1000.00. Tested sample on loan from manufacturer.

This is just a capsule review rather than a test report with measurements, as my exposure to the AV550 was somewhat limited. I inserted it into my home theater system right after having used the Marantz DP870/AV600 combination and found only improvements from a user standpoint, no disadvantages, despite the lower price. One reason was that the convenient on-screen displays are not disabled when the S-video connections are used for superior color performance. Another was the remote control that comes with the unit, the supremely versatile **Marantz RC2000 Mark II** learning remote (also available separately). I change audio and video components in my system too often to make the programming of an all-in-one remote worthwhile, but this one does it better than any other in my experience. Bass management is also quite sophisticated with the AV550.

The negatives, in case they matter to you, are a very rudimentary AM/FM tuner, no THX (just the Lucasfilm Cinema Re-EQ feature), no DTS, no tone-control bypass. For $1K this is not a bad deal.

Mono Power Amplifier

Marantz MA700

(Reviewed by Peter Aczel)

Marantz America, Inc., 440 Medinah Road, Roselle, IL 60172-2330. Voice: (630) 307-3100. Fax: (630) 307-2687. Web: www.marantzamerica.com. Model MA700 mono power amplifier, $500.00 each. Tested samples on loan from manufacturer.

In Issue No. 22,1 canonized the Marantz MA500 as the best moderately priced power amplifier known to me. The MA700 is similar in format—a self-contained single-channel amplifier on a deep and narrow chassis with a small frontal area—but bigger, more powerful, and not quite as conveniently slim. It costs 67% more; the question is, does it give you 67% more value? I think so, and David Rich seems to agree with me; he loves the circuit. He calls it "an excellent design with no tweako cultist practices." The circuit analysis that follows is David's; the measurements are mine.

An NPC2068 wired as a unity-gain buffer is at the input of the MA700. The first stage of the main power amplifier is a JFET-based differential pair with resistive loads. The JFETs are cascoded with bipolar devices. The second stage is another differential pair, this time all-bipolar with current-mirror loads. The differential devices are cascoded, but the current-mirror devices are not. Compensation is by dual Miller capacitors. The V_{BE} multiplier used to bias the output stage is part of this second gain stage. Two complementary pairs of bipolar emitter-follower predrivers drive the final bipolar output stage, which is composed of triply paralleled devices.

The amplifier is protected by a TA7317 device that monitors output voltage and current flow. This IC is part of an unusually comprehensive protection system. It is tied into a microcontroller that also monitors amplifier temperature, the presence of dc in the signal path, and output level. The micro can choose to reduce the power-supply voltage level to the output by activating a relay that switches transformer taps—or, if it gets really unhappy, it can pull the plug (literally open the relay in series with the power line). The protection system can also choose to mute the input signal or disconnect the speaker load from the amplifier.

One interesting feature of the MA700 is that the power supply for the voltage amplification section is taken from a voltage doubler circuit that uses the power supply of the output devices as the base voltage. This gives the voltage amplification section lots of headroom to work with, resulting in more linear operation.

The measurements confirmed the good impressions made by the circuit design. Frequency response deviations from dead flat at 1W/8Ω are -0.1 dB at 10 Hz, -0.07 dB at 20 kHz, and -0.5 dB at 85 kHz. Channel separation is, of course, virtually infinite with the monoblock contruction. THD + N into 8Ω. appears to be essentially noise-dominated at all frequencies, with minima varying between -94 dB and -97 dB just before the 200-watt clipping level. Into 4Ω, there is a bit of dynamic distortion; the 20 kHz minimum is -84 dB, whereas at lower frequencies the minima are in the -92 dB to -95 dB range, all immediately before the clipping level of 350 watts. I would call that awesome performance in a $500 power channel. I also checked the absolute noise floor with the input shorted; it was in the 5 μV to 75 μV range all the way up to 200 kHz, except in the two octaves from 10 Hz to 40 Hz, where it dropped to the 420 nV to 4 μV range. Again, I would call that quiet.

The PowerCube looked equally excellent, showing dynamic power of 318 W (50.4 V) into 8Ω/0°, 542 W (46.6 V) into 4Ω/0°, 827 W (40.7 V) into 2Ω/0°, and 793 W (28.2 V) into 1Ω/0°. Into ±30° and ±60° loads the output went slightly up in each and every instance, exactly as it should. What more can you ask for at this price?

It needs to be added that the amplifier is THX certified, has an input level control with a calibration mark for the THX reference level of 1 V, and is compatible with the Marantz remote control system (which requires a Marantz preamplifier or integrated amp).

David Rich points out that the topology of the MA700 is rather similar to that of the Marantz Model 15 of the 1960s (who else would have noticed?). Of course, there are significant differences—the Model 15 did not have the cascodes; its output transistor protection consisted of a lightbulb in series with the transistor's collector (hey, it worked!); and its output stage was a common-emitter-common-collector composite. It also employed what was an early but apparently effective approach to reducing crossover notch distortion. A seminal design of the legendary Sid Smith, the Model 15 became possible when Motorola introduced high-quality complementary bipolar output devices in 1966. Sid Smith, who had designed great tube amps, further proved his greatness as a designer by moving on to superior output devices as soon as they became available. He did not suffer from tube nostalgia. David Rich also reminds us that "the very forgettable Marantz Model 16 was designed after Smith had left the company." I must add that today's Marantz bears no relationship, in ownership or engineering staff, to the Marantz of the Sid Smith era and that the MA700 represents today's medium-priced state of the art as a result of a new and unrelated amplifier culture there.

Dolby Digital Processor

Marantz DP870

(Reviewed by Peter Aczel)

Marantz America, Inc., 440 Medinah Road, Roselle, IL 60172-2330. Voice: (630) 307-3100. Fax: (630) 307-2687. Web: www.marantzatnerica.com. Model DP870 Dolby Digital processor, $600.00. Tested sample on loan from manufacturer.

The Marantz AV600 nondigital AV surround preamplifier/tuner was reviewed in Issue No. 23 and is still in the Marantz line. The DP870 digital processor became available somewhat later to add Dolby Digital (AC-3) capability to the AV600 or any similarly configured AV equipment.

There is little to report here except that the DP870 is easy to set up and performs as intended. The difficulty of critiquing DSP algorithm implementation is explained elsewhere in this issue. I did, however, measure the DAC in the processor, as it is an important component of the digital system.

Through the optical input, the THD + N measurements at 44.1/16 across the audio spectrum indicate from 13 dB to 23 dB excess distortion at full scale above the theoretical minimum of-98.08 dB. That's awful, and not all of it is gain-related analog distortion because at no frequency can the excess distortion be lowered to less than 10.6 dB simply by reducing the digital input. What we have here is cost cutting on the DACs. Gain linearity is not too bad but no more than fair: 0 dB error at -70 dB, +0.25 dB error at -80 dB, +0.6 dB error at -90 dB. Even the frequency response through the optical input droops to -0.13 dB at 10 kHz and -0.4 dB at 20 kHz.

Don't misunderstand the above figures, however. I seriously doubt that they impose any limitation on the transparency of the audio, which in AV country is almost certainly dependent on the DSP algorithms.

Line-Level Preamplifier

Morrison ELAD

(Reviewed by Peter Aczel)

Morrison Audio, 334 King Street East, Toronto, Ont., Canada M5A 1K8. Voice: (416) 362-0523.. Web: www. surpher.com/MORRISON. ELAD line-level preamplifier, $590.00 (direct from Morrison). Tested sample on loan from manufacturer.

What is the worst nightmare of the high-end audio fraternity? A clearly better audio product at a much lower price. Yes, there is a story here, with a moral of great poignancy.

The story began in Massachusetts in 1992, at the laboratories of Analog Devices, where a gifted engineer by the name of Scott Wurcer had just developed a new IC op-amp featuring phenomenally low distortion and noise at audio bandwidths. David Rich, in Issue No. 18, almost immediately called the audiophile community's attention to this veritable God's gift to the preamplifier designer, the AD797. And what happened? Little or nothing. I don't know how many AD797's were sold to manufacturers of professional audio equipment (microphone preamps, mixing consoles, etc.), but I do know that the consumer audio industry responded with a big yawn. I became aware of one application, by Theta, in an otherwise undistinguished midpriced outboard DAC, but that was about it. The Krells and Mark Levinsons and Audio Researches of the high-end mafia obviously didn't want to have anything to do with a lowly op-amp chip, even if it could do circles around their golden-ear-approved topologies. Why do it the easy way when the hard way works so well for us?

Years passed. I suppose Analog Devices found a market for the AD797 because it remained in the line. Then, finally! Circa 1996-97, Don Morrison of one-man Morrison Audio, located in the area that is the Florence of today's Canadian audio Renaissance, had an inspired idea. Let's design a simple line-level preamp/control unit around the AD797 and freak out the high-end cultists. (The idiocies of the latter are the subject of Don's favorite anecdotes—see also "Box 978" in this issue—so I think I can get away with ascribing to him such a motivation.) The result was higher-than-high-end performance at a mid-fi price. The problem that remained was credibility. How can a Hyundai be better than a Lamborghini? Well, in the delirious world of consumer audio, it *can* be and in this case *is.*

Of course, the only way I can deal with the credibility issue is to give you my test results. They won't convince the brainwashed ignoramuses who believe that the best-sounding equipment can never be the one with the best measurements, but then it has always been difficult to fight voodoo with science. Here are the measurements:

Frequency response, at 1 V output, 10 Hz to 70 kHz, ±0.0 dB, rolling off to -0.2 dB at 200 kHz (I cannot measure accurately below or above that band). Channel separation, at 1 V output, 105 dB or better from 20 Hz to 2 kHz and 98 dB or better from 2 kHz to 20 kHz. Noise floor, with inputs shorted and gain at maximum, 0.2 μV or lower from 10 Hz to 1 kHz, 0.8 μV or lower from 1 kHz to 20 kHz, 3.3 μV or lower from 20 kHz to 200 kHz. Best of all, THD + N at the maximum output of 10 V is -100 dB (0.0010%) at nearly all frequencies, rising to -97 dB (0.0014%) at 20 kHz. Into a 600-ohm load maximum output drops to 9 V, with no other change in performance. These figures have never been surpassed individually in my testing experience and never equaled in combination. The Morrison ELAD is the most nearly perfect line-level preamp known to me.

Now, it must be pointed out that the unit has two volume controls, one for each channel, in lieu of a separate balance control, and that definitely helps the channel separation. There are only two inputs, selected by a toggle switch; a second toggle switch mutes the two available outputs. That's about all there is to the preamp—minimalism at its most intense. There are two small genuflections to the blessed high-end sacraments, however. One is the separate power-supply chassis; the other is the absence of a power on/off switch. To me only the latter is a minor irritation, since the sum of the two chassis is still smaller than one conventional preamp. That separate small power-supply chassis is actually quite heavy as a result of serious overdesign, and the metalwork is also of a heavy gauge, so that the visual impact is "rather agricultural, don't you know," as a London audio salesman once remarked to me about some other equipment. Compact agricultural, that is. The rugged XLR plugs on the power-supply cable reinforce the impression.

One interesting feature of the preamp is that the circuitry designed around the AD797's is potted, to foil the untutored modifiers. There is some careful engineering in there, including a second stage of regulation, and the extraordinary performance is almost certain to go to pot in the wake of some know-it-all tweak's soldering iron.

What do you give up when you pay only $590 for a state-of-the-art line stage? Remote control, for one thing (a TV carryover, no big deal in audio). Lots of inputs, for another, but an inexpensive outboard switching box can take care of that. And, of course, home-theater AV controls, which are another ballgame altogether. Not much else, really. Compare with the $15K Conrad-Johnson ART tube line stage and weep. The C-J's measurable performance is almost obscenely inferior. Only the *sancta simplicitas* of the faithful makes it "sound better" and worth the 25½ times higher price.

The poignant moral I referred to at the beginning? To put it as delicately as I can without trivializing the basic issue: the designers of megabuck line-stage circuitry are playing with themselves and their marketeers are playing with *you.*

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David Rich analyzed the circuit topology of the ELAD and sent me the following memo:

"The AD797 is wired for 6 dB gain. The amplifier is in the noninverting gain configuration with a dc servo circuit formed with a very low offset OP177. No capacitors are in the signal path. I am somewhat worried about volume-control noise occurring in the future because the bipolar AD797 does draw dc bias current. A small input coupling cap would have solved the problem. Secondary regulation is in the box with LT317 regulators. They bring the externally applied ±22.4 V supplies down to ±16.7 V. Near the power supply is another pair of LT317 regulators. The unregulated rails are ±27 V. Overall an excellent design which achieves excellent measured results."

Outboard D/A Converter

Parasound D/AC-2000

(Reviewed by Peter Aczel)

Parasound Products, Inc., 950 Battery Street, San Francisco, CA 94111. Voice: (415) 397-7100. Fax: (415) 397-0144. E-mail: sales@parasound.com. Web: www.parasound.com. D/AC-2000 outboard D/A converter, $1995.00. Tested sample on loan from manufacturer.

Until now, for some strange reason, we never had a piece of equipment with an UltraAnalog converter in it come our way. This is the first one. David Rich, in Issue No. 15, pointed out that the UltraAnalog DAC, a hybrid circuit, had the lowest guaranteed THD levels of any D/A converter. That was eight years ago, but the pecking order has remained pretty much the same, at least as far as Parasound is concerned. The D/AC-2000 is Parasound's "statement" product for the audiophile who can't live without a multi-input D/A processor on a separate chassis.

The sleek black box comes with four digital inputs (coaxial, AES/EBU, Toslink, ST optical) and offers both balanced and unbalanced outputs. The front panel has LEDs to indicate sampling frequency (32 kHz, 44.1 kHz, 48 kHz), HDCD decoding, polarity (right above an invert switch), and de-emphasis, in addition to the four input indicator LEDs. Can't ask for much more.

As for the circuit, the S/PDIF decoder is the UltraAnalog AES C003 module, the digital filter is the PMD-100 from Pacific Microsonics, and the D/A converter is the UltraAnalog DAC D20400. All high-end icons. David Rich's circuit analysis tells me that the S/PDIF decoder gets its own ±12 V supply and that each analog section also gets its own ±12 V power rails. The DAC has provisions to be powered by two sets of power supplies, and Parasound powers it with the two analog supplies. The analog section's power comes from its own transformer, with 6600 µF of capacitance on the unregulated rails. The digital section is powered by a separate transformer and has two 5 V regulators. The 6600 µF of capacitance at the bridge rectifier of the digital power supply is surprisingly large, says David. The S/PDIF decoder, digital filter, and DAC get one of the digital supplies; all other digital circuits get the other. The switching of digital input signals is done by relays rather than logic gates, an expensive approach which must be perceived as a method to reduce jitter. All regulators are in the 78*xx*CT and 79*xx*CT family-

The analog stage starts with a passive 3rd-order elliptic filter, complete with series inductor ("what will the tweaks think of this?" asks David). The active analog stage is formed with the Analog Devices OP275 and an added complementary follower output stage. Separate inverting and noninverting stages are provided to form the fully balanced output. A relay mutes the output, and a 3rd-order passive network is at the output to remove even more RFI energy. David believes this is clearly in response to EEC rules. It is interesting, he observes, that the FCC has been relatively lax on this issue, but the Europeans seem to be really enforcing their laws.

The construction of the Parasound is very good. The sheet metal is of high quality, and the double-sided PC board is stuffed with high-grade components. David would like to substitute an op-amp of higher performance, such as the AD797, and goose the analog supply rails up to ±15 V.

Needless to say, I was pretty eager to measure the distortion performance of the fabled UltraAnalog converter. I was not disappointed. Instead of deluging you with endless numbers, let me keep things very simple just by specifying the deviation from theoretical perfection, which is -98.08 dB THD + N when the word length is 16 bits. The Parasound comes within 0.5 dB of that, at all frequencies, when the digital input is well below full scale. With a full-scale input, gain-related analog distortion enters the picture, but even then the excess distortion is only 2.4 dB. (With an AD797 and ±15 V analog supply rails, as David suggests, the 0.5 dB figure would probably hold even at full scale.) This is as close to texbook-perfect performance as I have seen, but I must confess that these figures apply only to the better channel in my sample; the other channel was worse by 2.5 to 3 dB. With a 20-bit word length I measured only about 6 dB lower THD + N in the better channel, which is more like 17-bit performance—but that's exactly what I expected.

Gain linearity with the UltraAnalog converter is as close perfection as with any delta-sigma DAC I have measured: 0 dB error down to the -90 dB level, no more than 0.25 dB error down to -110 dB. Wow! The "Rob Watts test" (FFT spectrum of a dithered 1 kHz tone at -60 dB) shows a bin-by-bin noise floor of -126 dB and no harmonic blips whatsoever. Channel separation is outstanding: 117 dB or better at any frequency under 1 kHz, still 107 dB at 20 kHz.

One little peculiarity: the frequency response at full scale droops by 0.18 dB at 10 kHz and 0.65 dB at 20 kHz. Could that be some kind of audiophile tweak? I have no idea.

Be that as it may, I am unaware of any more nearly perfect digital equipment than the Parasound D/AC-2000.

5-Channel Power Amplifier

Rotel RB-985 THX

(Reviewed by Peter Aczel)

Rotel of America, Equity International, Inc., 54 Concord Street, North Reading, MA 01864-2699. Voice: (978) 664-3820. Fax: (978) 664-4109. Web: www.rotel.com. RB-985 THX 5-channel power amplifier, $1000.00. Tested sample on loan from manufacturer.

Build quality, topology, measurements. That's all there is to the evaluation of a power amp. There is no mystique, no gestalt, untutored subjective reviewers notwithstanding.

The construction of this plain black box is typical Rotel. Definitely better than Japanese mainstream but not as good as Bryston or Mcintosh. The sheet metal is of good quality. No push-on connectors are used inside. Resistors are all metal-film. On the other hand, the small primary power-supply filter capacitors are inimical to good THD performance at the lowest frequencies. They have insufficient energy storage for keeping the power-supply line ripples to a low enough level. As David Rich points out, this is an effect we seldom see in modern power amplifiers and certainly not at this price point.

As for circuit design, basically the same complementary topology can be seen in the 5-channel RB-985 as in the 2-channel RB-990BX reviewed in Issue No. 20. The dual output devices are driven by two stages of emitter followers. Both amplifiers lack emitter followers between the first and second voltage-gain stages; that lack of isolation of the two stages from each other explains the measured dynamic distortion, according to David Rich. The very similar but much costlier Rotel RHB-10 (see Issues No. 21 and 22) includes the extra stage and has considerably lower dynamic distortion. Another small problem of the RB-985 is a current-limit sensor that appears to be a bit touchy, as witnessed by the PowerCube results.

The measurements came out as follows. Frequency response at 1W/8Ω: -0. 1 dB at 10 Hz, -0.1 dB at 20 kHz, -0.45 dB at 50 kHz. Front left/right channel separation: minimum 75 dB (in the 5 to 10 kHz octave), maximum 100 dB (at the lowest frequencies). THD + N into 8Ω: basically noise-dominated, 1 kHz minimum -91 dB, 20 Hz minimum -81 dB (cause discussed above above), 20 kHz minimum -75 dB (dynamic distortion), all just before clipping at 150 W. THD + N into 4Ω: again basically noise-dominated, 1 kHz minimum -87 dB, 20 Hz minimum -77 dB (same thing!), 20 kHz minimum -70 dB (dynamic distortion), all just before clipping at 240 W. The noise floor with an 8Ω load and inputs shorted fluctuates between 5 μV and 15 μV over the audio range, with some minor 60 Hz and 120 Hz peaks.

Into resistive (0°) loads of 8Ω/4Ω/2Ω, the PowerCube system measured 200W/350W/339W of dynamic power (i.e., 40V/37.4V/26V). Into the corresponding reactive loads, the dynamic power dropped slightly at the 2Ω/±30°/±60° test points but was OK at all the others. As for 1Ω, either resistive or reactive—forget it. The amplifier cries uncle and shuts down.

Now these are quite decent results on the whole, but they could be significantly better with some very minor changes, such as larger power-supply filter capacitors, the slightly more complex RHB-10 topology, and a more sophisticated protection circuit. Come on, Rotel. Give Bryston and Mcintosh something to worry about.

Compact Disc Player

Sony CDP-XA20ES

(Reviewed by Peter Aczel)

Sony Electronics, Inc., 1 Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 358-4060. Web: www.sony.com. CDP-XA20ES compact disc player with remote control, $750.00. Tested sample on loan from manufacturer.

The $3000 CDP-XA7ES is still Sony's flagship CD player and my top choice when price is no object. (See Issue No. 23.) The XA20ES is an attempt by Sony to make a comparably strong statement with a unit costing ¼ as much. The attempt must be called at least a partial success, since the quality/performance difference between the two players is much smaller than the price difference. One feature that Sony is trying to use as a high-end signature to tie the two products together in the mind of the consumer is the unorthodox disc transport. In both models the laser pickup is fixed and the rotating disc is carried past it on a moving sled. The mechanism is a little less robust in the XA20ES; the slightly annoying disc stabilizer (mislay it—no play) is smaller and lighter; and the read circuitry is similar but not identical. David Rich prefers the old Sony G-base transport assembly of the discontinued X779ES to either version but points out that the overall build quality of the XA20ES is still well above that of standard Japanese audio components.

The digital electronics are functionally similar in the senior and junior models but the chips are not the same. A single chip combines the digital filter and noise-shaped DAC in the XA20ES; it appears to be functionally equivalent to the corresponding two chips in the XA7ES, but we "do not have details on FIR tap lengths, multiplier, data path, and coefficient word lengths to make a really informed judgment," says David Rich.

David finds the analog circuitry in the XA20ES to be a comedown not only from the XA7ES but also—and perhaps more significantly—from previous Sony CD players at this price point. The highly sophisticated discrete regulator for the analog side of the D/A is gone. Small 7807L and 7907L regulators replace it for the analog supplies, and a 7805L replaces it on the digital side. The analog supply rails are a low ±7 V. The four subregulators used on the XA7ES are gone, as is the independent crystal oscillator and its subregulator. The XA20ES uses the oscillator on the digital filter IC. This will result in a clock with more phase noise. The dual transformers of the XA7ES are, not surprisingly, replaced by one, but that single transformer of the XA20ES is not a cheap one, as it has four secondaries. Supply capacitors on the analog side have been dropped from 4700 µF to 3300 µF. The digital side drops by half to a still substantial 6800 µF

The analog signal path itself follows the same basic design in the two players, but in the XA20ES the M5238P and NJM2114 op-amps have replaced the AD712 units. The discrete output stage and the dc servo used in the final stage of the XA7ES are also gone. The signal flow in both players goes from two single-sided 3rd-order Sallen-and-Key filters (one for each of the two differential outputs of the filter) to a differencing amplifier with a lst-order low-pass response and finally to another 2nd-order Sallen-and-Key filter. This fixed-level signal is sent to a volume control to provide the option of a variable output, but the buffer amp that follows the control in the XA7ES is missing from the XA20ES, nor are the balanced outputs of the XA7ES available on the XA20ES—that whole circuit is gone with the wind of simplification.

Now then, how does all that affect the measurements? (Did I hear some Harleyfied naïf ask how it affects the sound? You need to read some of our back issues, guy.) Basically, the XA20ES is a good clean CD player, not quite the equal of the XA7ES in full-scale D/A performance at the higher frequencies but otherwise quite comparable to it and—this is a surprise—actually superior in digital error correction. On both the old Pierre Verany and the new Digital Recordings test discs, the lower-priced player did a little better on the error-correction torture tests than the flagship. Of course, both passed the tests that are within the CD standard and some steps beyond.

The full-scale (0 dB) THD + N of the XA20ES is within 1 dB of the theoretical 16-bit limit of -98.08 dB at the lower frequencies. The excess distortion reaches 2 dB at 1 kHz and skyrockets to 17 dB at 10 kHz. (It gets even worse if the test bandwidth is widened to 80 kHz, but that's not considered realistic by most practitioners.) What's happening here is gain-related analog distortion, as indicated by the return to 1 dB excess distortion, or less, across the entire audio range when the digital signal level is reduced to -24 dB. The XA7ES, as my review in Issue No. 23 stated, did not exhibit more than 1 to 2 dB excess distortion at any digital level regardless of frequency. Its analog circuitry is clearly superior to that of the XA20ES.

In other respects the XA20ES is pretty close to perfection. Quantization noise: -97.9 dB. Dynamic range: 97.9 dB. Channel separation: 102 dB at 16 kHz, increasing to 124 dB at low frequencies. Frequency response: +0.0/-0.1 dB from 10 Hz to 20 kHz. Low-level linearity error: 0 dB all the way down to -100 dB. FFT spectrum of a dithered 1 kHz tone at -70.31 dB (modified "Rob Watts test"): no harmonic blips whatsoever, bin-by-bin noise floor of -127 dB. Pulse response: positive polarity. Monotonicity test pattern: slight glitches but no serious anomalies.

My conclusion is that the Sony CDP-XA20ES very neatly splits the difference between ultrahigh-end and standard midpriced CD players. It would be even more attractive if it had been designed with late-'80s/early-'90s attention to quality details.

CD/DVD Player

Sony DVP-S7000

(Reviewed by Peter Aczel)

Sony Electronics, Inc., 1 Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 358-4060. Web: www.sony.com. DVP-S7000 CD/DVD player with RMT-D100A remote control, $1199.00. Tested sample on loan from manufacturer.

This was Sony's original all-out, reference-quality DVD player, released a little too late to be reviewed in our last issue and now, in consequence of our dilatory publishing schedule, a little too old to be reviewed as an exciting new item. It was, however, still in the Sony line when I last checked, even if labeled "first generation" by reviewers now that "third generation" is the marketing buzzword. (The new DVP-S7700, with 96/24 DACs and all sorts of other updates, is scheduled to replace it for $200 more.)

The remarkable thing is that Sony was so far ahead of the curve when they came out with the DVP-S7000, and so intent on including all possible goodies available at the time, that I experience absolutely no hankering after the "third generation" when I view a DVD movie on this equipment.

It must be immediately pointed out that the video quality of DVD playback depends not only on the player but also on the disc, the decoder/processor, and the TV monitor, and I have neither the inclination nor the test setup to separate those four factors subjectively. I can make the sweeping statement, however, that the DVDs of recently made motion pictures are distinctly superior in video quality to any videocassette or laser videodisc played through the same high-end home-theater system in my home. Is that due to the specific features of the DVP-S7000 or the overall superiority of the DVD technology? I shall have more authoritative answers to such questions when we bring our video bench-testing protocol up to the level of our various audio protocols and do comparative evaluations of DVD players such as we have done with preamps and power amps. (At this point I could not even evaluate the component-video output of the Sony, having only coax and S-video inputs on my aging monitor.) Meanwhile here are some basic measurements of this player's CD performance, which Sony originally announced to be right up there with their best CD-only players. Not quite so.

The frequency response at full scale is ruler flat from 20 Hz to 15 kHz and -0.1 dB at 10 Hz and 20 kHz. THD + N at full scale is approximately 8 dB higher across the audio band than the theoretical limit of-98.08 dB (for 16 bits). I'd like to report that this is gain-related analog distortion, but it isn't. With the digital input reduced to -24 dBFS, there is still 5.7 dB distortion in excess of the adjusted theoretical minimum—and that comes from the DAC, almost certainly. Sony's CD-only players don't do that. Yes, the gain linearity is absolutely perfect down to the lowest levels, even -100 dB, but I expected that from the Sony "current pulse" D/A conversion system. Monotonicity is OK with very minor aberrations; de-emphasis error is zero; channel separation is in the 100 dB to 134 dB range (depending on frequency and the driven channel); polarity is noninverting; error correction (according to the new Digital Recordings test disc) is about as good as that of the Sony CDP-XA20ES. All in all, a decent result.

A few small quibbles. The menus could be a little more user-friendly. The default setting for the digital output is PCM, not Dolby Digital (AC-3)—it should be the reverse, since the player can be assumed to have been purchased by the consumer mainly for playing movies. If there is a temporary power failure or the unit is unplugged for some reason, the 5.1 surround sound is lost until the user remembers to reprogram. Even more annoying is the tendency of the player—at least of my review sample—to turn itself off in the middle of a program, for no reason. This happened to me three or four times, with different DVDs, and I could not figure out what triggered it. The remote control is OK but I have seen ergonomically better ones.

None of the above imperfections is serious enough to be an argument against the basic design. The Sony DVP-S7000 was state-of-the-art when it debuted and remains a fine machine.

Hi-Fi VHS VCR

Sony SLV-M20HF

(Reviewed by David Rich)

Sony Electronics, Inc., 1 Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 358-4060. Web: www.sony.com. SLV-M20HFHi-Fi VHS videocassette recorder, $499.99. Tested sample owned by reviewer.

This is absolutely the last item to go into this greatly delayed issue (the Editor is ready to kill me), but I just have to tell you about this VCR, which is unlike all others. More precisely, it is almost exactly like all others (see *Consumer Reports* for typical examples) except for two very significant differences. One is that it uses the **Gemstar Guide Plus+** system. This is basically similar to the old StarSight Telecast system (Gemstar now owns StarSight), but it is free. (Little banner advertisements appear at the side of the guide display. These pay the cost of the system.)

If you are unfamiliar with StarSight, what it and Gemstar do is to give you a complete programming grid for all covered stations, running for a number of days (in this instance, two days). Click on any block in the grid and you can find out more details about the show. Click on another button and you set up the VCR for taping. Yes, one click does it! No more VCR guide numbers to type in or manual programming menus to fill in. One click does it. Of course, the signals you are interested in must also be of interest to Gemstar. They do not cover my local (Lehigh Valley) PBS station, nor do they cover the Ovation network or BBC America. Gemstar claims the number of stations that can be covered is limited by the data rate at which data can be sent. If you try to complain that they must start carrying your favorite station at the expense of some old movie channel, you are told that since the service is free you have no leverage on them.

Unlike the old StarSight system that placed the digital data signal (in the vertical blanking portion of the TV signals) on PBS carriers, the Gemstar system uses common cable stations. Many PBS station were not carrying the StarSight signal, rendering the system inoperative in those areas. By determining where on the dial the Gemstar-carrying cable signals appear, as well as using time cues, the SLV-M20HF Guide Plus+ system figures out where you live and what cable carrier you are using. With this set of information, the Guide Plus+ system figures out what data should be downloaded to the VCR and what channels match which network. No six-year-old computer genius is required—your grandmother could set up the SLV-M20HF because the incredible complexity of the system is transparent to the user.

On top of the Gemstar guide, Sony also throws in **SmartFile,** the second big difference. This is a strip that you put on the back of a VCR tape. It contains an inductively powered and coupled nonvolatile memory circuit that stores data on what is recorded on the tape. That data comes from (you guessed it) the Gemstar Guide Plus+. Of course, if Gemstar does not track the station, all you get is the recording time and the station number—which is why you do not want to purchase the $50 cheaper version of the SLV-M10HF, which has SmartFile but not the Gemstar Guide Plus+ system. You do not need to put the tape in the VCR to view the SmartFile data—just hold the tape near the inductive coupling pod on the VCR and the data appears on the screen. The days of unmarked VCR tapes are gone forever. Unfortunately, the SmartFile strips cost $2.00 each (more than the VCR tape!). The price is not out of line with the technology used but it clearly limits the use of the system. It is not out of the question that SmartFile could be the El Cassette of the VCR world. A large price drop for the strips and the use of the technology by companies other than Sony would be a sign that SmartFile is here to stay.

The days of recording the wrong program are now over. (Yes, even Ph.D. E.E.s do it more often then we will admit to.) Long live the Gemstar Guide Plus+ system and its embodiment in the Sony SLV-M20HF!

A V Surround Receiver

Sony STR-DA80ES

(Reviewed by Peter Aczel and David Rich)

Sony Electronics, Inc., 1 Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 358-4060. Web: www.sony.com. STR-DA80ES Dolby Digital receiver, $1200.00. Tested sample on loan from manufacturer.

This is basically the same AV receiver as the top-of-the-line Sony STR-DA90ESG, but for $400.00 less. What you give up are the on-screen readouts (to me that's not trivial), adjustable DSP parameters, some EQ facilities, and DSS (satellite dish) receiver control. The core hardware is all the same.

The receiver is remarkably well-designed, especially from a per-dollar perspective, and will accommodate more analog and digital program sources than one would expect even in a fairly elaborate AV home system. There is a very sexy motor-driven lid that covers/uncovers all the secondary controls, leaving only the essentials exposed. There are more surround and virtual-surround settings than I would ever want to use, all clearly indicated on the front-panel dot-matrix display (as for me, I'm strictly a Stereo/Dolby Pro Logic/Dolby Digital person). Let me cut straight to the measurements then, of which I made only some very basic ones, as the receiver was available to me only for a relatively short time.

The power amplifiers in the unit incorporate an 8Ω/4Ω. impedance selector, for the same reason as the Yamaha AX-592 integrated amplifier reviewed by David Rich in this issue. (See his explanation; it's basically a safety device in case the load impedances get crazy low. The 8Ω setting yields the better numbers in all tests.) The five power amps in the Sony are distinguished by excellent current capability, but they suffer from considerable dynamic distortion. Into 8Ω. in direct mode, the THD + N curves are identical (and completely noise-dominated) for 20 Hz and 1 kHz signals, bottoming out at -87 dB just before clipping at 110 watts. The 20 kHz curve parts company from the others at 200 mW (!) and reaches a minimum of only -68 dB shortly before clipping at 110 watts. The distortion curves have almost the identical profiles when the impedance selector is switched to 4Ω. and a 4Ω load is connected, the only difference being 5 to 6 dB more distortion all around. Small-signal frequency response is -0.2 dB at 20 Hz, -0.1 dB at 50 kHz, and dead flat in between; front left/right channel separation is 40 dB at 20 kHz (not so hot), increasing linearly to 87 dB at 20 Hz (better).

The PowerCube looks very good indeed with the impedance selector at 8Ω. Dynamic power into resistive (0°) loads of 8Ω/4Ω/2Ω/1Ω is 145W/238W/341W/384W (34V/31V/26V/19.6V), showing quite remarkable current capability for a midpriced receiver. Into reactive loads the output is slightly higher at each test point, which is the desired result.

The D/A converter measurements show impossibly high gain-related analog distortion at full scale, but with the digital input reduced the excess distortion relative to the theoretical 16-bit minimum is an acceptable 6 dB. Gain linearity, on the other hand, is virtually perfect all the way down to -100 dB, where the error is still only ±0.3 dB. And that's all she wrote.

—***Peter Aczel***

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What we have here is a remarkable piece of engineering, given its low price. Judging how well the design has been implemented is impossible, however, with the testing methodology I use. The problem is that most of the design issues are embedded in software in one of the three (count them, 3) DSPs that this thing uses. On top of that you get two microcontrollers and three custom ASIC devices that perform the digital interpolation and delta-sigma D/A conversion. Each ASIC handles two channels. Now, the performance of this very complex system comes down to the nature of (1) the digital hardware—data path sizes, coefficient sizes, multiplier rounding and truncation methods, proper use of dither, etc.—and (2) the digital software—are the algorithms robust against clipping? are there any limit cycles or other noises generated due to improper coefficient selection? is nonlinearity introduced as a result of coding errors? have the digital algorithms for perceptual decoding been properly implemented against the Dolby AC-3 standard? etc.

Now there are two ways to determine if this has all been done correctly. One is to have the manufacturer supply complete details on the DSPs and ASICs along with the source code for the DSPs. Now, even if the manufacturer were crazy enough to do this core dump of IP (intellectual property) for our use, there is no way we could interpret the information (ever try to slog through raw DSP assembly code?) and check for design errors in a reasonable period of time. That brings us to the second possible approach, which is to develop novel test signals and protocols to uncover hardware and firmware design errors or cost reductions. Since AV systems have a very low priority within my personal range of interests, you are going to have to look somewhere else for these new tests to be developed. Luckily, David Ranada of *Stereo Review* appears to be up to this challenge, and his reviews are definitely must reads.

One nice thing about all the digital processing is that it gets rid of a lot of analog complexity. Tone controls, bass crossover filters, and bass management all happen in the digital domain. In the future I predict that everything will happen there, with just a small class D amplifier (or some variant of pulse technology, such as delta-sigma modulation) representing the only analog left. It is possible it could all be boiled down to a pair of switching transistors per channel and a passive output filter. For now there is still some analog to talk about. First, there is a phono stage (many AV receivers leave that out) based on an M5218 op-amp. The M5218 is also used in the rest of the analog chain of this unit. You also get the cheapest tuner Sony knows how to make. They are too embarrassed to use it in Europe, where they still care about FM, so they remove it and sell this as the TA-V88ES integrated amp over there.

Analog input switching is by way of a Sanyo CMOS chip. Latchup is prevented the el cheapo way with series 1k resistors. Analog signals get converted by a CXD8681 A/D converter. Since this is an internal Sony design, its operation is a mystery to me as is the CXD8505 interpolator and DAC chip that performs the inverse operation. The differential output of the DAC gets converted to a single-ended signal by the use of three op-amp sections. Reconstruction filtering gets done with another op-amp section. All six signals next go through another op-amp that functions as a 3-bit volume control. The passive part of this circuit is a discrete affair that is formed with a resistor ladder and bipolar switches. Everything but left and right also gets finer level adjustments by way of a Sanyo LC7553 integrated potentiometer. One assumes this is for making relative level adjustments. The left- and right-channel signals pass through a relay which selects between the DACs or the analog signal that came off the Sanyo CMOS input selector. When you select the analog path, the .1 channel may be deactivated (it is not clear from the info I have), thus this mode may be a full-range-speaker-only mode.

All the signals next go into a six-gang (ugh, what mistracking, what reliability problems!) motorized potentiometer that was old technology before I was born. We assume this pot is in the signal path because Sony has not figured out how to do low-distortion digital volume controls (as Denon has) or, more likely, they know that is a more expensive option so they went with cheap, unreliable old tech.

New tech is found in the power amp, which consists of a monolithic voltage-gain-stage IC. This is not such a bad thing even if its main purpose is to save space and cost. The differential-pair front stage gets a real current source and active current-mirror loads. The second gain stage consists of a cascoded common-emitter stage with degeneration and even a cascode stage! This stage comes out of the chip to drive the output stages through a discrete V_{BE} multiplier and then back on the chip to be terminated into a current source. The healthy (yes, good PowerCube!) output stage is also formed with composite structures. All the output devices and predrivers for the *npn* side are in a single package. The *pnp* side is similar. All stages have discrete current-monitoring circuits, which work quite well judging from the PowerCube. The higher-than-normal dynamic distortion may be the result of slow lateral *pnp* devices in the composite voltage-gain stage, or it could be the result of second-order effects that occur when so much circuitry is squeezed into such a small space. Although the overall performance of the DACs and analog components is not state-of-the-art, the numbers are clearly at least an order of magnitude inside the plane of audibility. The power amp can drive difficult loads as well as anything, up to its power limit.Six channels of this near state-of-the-art stuff, three DSPs, two microcontrollers, and a nice user interface for $1200—isn't progress wonderful? If only I were an AV enthusiast...

—David Rich

Indoor/Outdoor FM Antenna

Terk FM Pro FM-50

(Reviewed by David Rich)

Terk Technologies Corp., 63 Mall Drive, Commack, NY 11803. Voice: (516) 543-1900 or (800) 942-8375. Fax: (516) 543-8088. E-mail: terk@pipeline.com. Web: www.terk.com. FM Pro FM-50 indoor/outdoor FM antenna with Power Injector, $119.95. Tested sample on loan from manufacturer.

This is a ½-wave dipole in a plastic box that is suitable for hanging indoors or out. Also in the box is a well-designed broadband amplifier with a gain of 11 dB, which is completely bypassed with a remotely controlled relay. The disadvantage of a dipole is that it is directional; thus you have to aim the antenna. Your walls may or may not be in the optimum position, and if you have signals coming from multiple directions you have an even bigger problem. Hanging it from the ceiling so it can be rotated is a solution but not one with a high WAF (Wife Approval Factor).

The advantage of a dipole is that it will receive horizontally polarized signals. FM stations are generally circularly polarized, so that both horizontally and vertically polarized antennas may receive the signal. Sometimes the power varies between the signals. For example, during peak driving hours more power may be sent into the vertical component so cars can get the signal clearly. From the standpoint of DX-ing, an antenna receiving a horizontally polarized signal is the best bet, since vertically polarized signals attenuate at a faster rate. This is a result of razor-edge diffraction effects, according to Richard Modafferi.

A small remote box to be connected at the receiver end powers the antenna amplifier through standard coax cable. Terk supplies a standard balun for receivers that have only 300Ω inputs. In addition, they provide a passive coax-to-bare-wire converter for el chepo receivers that have 75Ω. inputs using push-in connectors (like the Pioneer receiver we reviewed in the last issue). This is the first time I have seen this very useful adapter. Clearly Terk wants its antenna to perform optimally and is not going to be prevented by cost-cutting at a receiver's terminals.

So how does it work? In my experience it was equal to or better than any other indoor antenna I had available, provided it was correctly aimed. The AudioPrism APPA-8500 was much more convent to use, since its directionality patterns are remotely controllable, but signal levels were weaker on distant stations, since it is vertically polarized. Richard Modafferi reports similar results, calling the Terk the best FM antenna he has tested so far. It even beat his own ¾-wave indoor vertical. He reports that a distant signal on 91.3 MHz (WCNY) was usable with the Terk antenna, with the internal amplifier producing slightly better results. Richard's vertically polarized antenna could not bring in a usable signal unless the Magnum Dynalab amplifier was added to the chain.

I was amazed to find that I did not have a spuriae problem with the untuned amplifier switched in, as I have a 50 kW station a few miles away. Richard did find some spuriae from cell-phone and pager operators that have a tower in his backyard. Clearly a worst-case test. Should you have spuriae problems, you can always switch the amplifier out of the signal path with the remote relay bypass. Other antennas do not bypass the amplifier but just reduce its gain, so the amplifier is left in the signal path and can still contaminate the signal.

By far the most amazing signal improvements occurred with the dirt cheap tuner in the Pioneer receiver after I had removed a small tunable amplified Radio Shack antenna (12-1833) and replaced it with the Terk. A number of unusable signals now emerged cleanly. If you are having reception problems with an indoor antenna, replacing it with the Terk is clearly the most economical first step, provided you can deal with the aiming problem.

Terk also makes an AM antenna that I also had a chance to test. This one is tunable. It is basically a thin 6-inch circular ring mounted on a flat stand. I did not subject it to same rigorous set of tests that we did with the FM model, but I can report it improves AM reception significantly over smaller antennas supplied with tuners. I can join Richard in talking about receiving WQED from New York City 100 miles away with this antenna attached to nothing better than the Pioneer receiver. Again a very impressive result from Terk. Clearly Terk's decision to increase the size of its products has resulted in significant performance improvements. I highly recommend that you try these antennas if you are having reception problems.

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...and something you always looked for:

Service Manuals

A. G. Tannenbaum

A. G. Tannenbaum, P.O. Box 386, Ambler, PA 19002. Voice: (215) 540-8055. Fax: (215) 540-8327. Web: www.agtannenbaum.com. Service manuals for discontinued equipment.

What do you do when you need a service manual and the manufacturer no longer has it in print? One source is Sams, but their audio selection is limited and the quality of the reproductions is not so hot.

A better source is A. G. Tannenbaum. Their Web site is www.agtannenbaum.com. Copies are lovingly reproduced in a binder format. No more loose oversized pages. No more unreadable photos, PC board traces, and schematics. Prices are low and selection is very wide.

You may want to, as I did, purchase manuals to examine classic designs, even if you do not own the units. Early Marantz manuals make for good reading, as do Kenwood top-of-the-line tuners from the '70s. Yes, you can even look deep into the past and see why early Fisher, Sherwood, and Scott designs were not transparent.

—David Rich

Truth can never be told so as to be understood, and not be believ'd.
—WILLIAM BLAKE (1757-1827)

New World Cyborgs

By Tom Nousaine
(The High-Definition Weasel)

I love Golden-Ear reactions to bad news. Over the past year or so Golden-Ear Apologists have started labeling anybody who uses or accepts results of controlled listening tests as a 'Borg. Scientists like Jim Johnson of AT&T and Bob Myers of Hewlett-Packard are depicted as Cyborgs or machine-like beings that do not ever listen to music. Kind of like when Ken Kessler called me a "propeller-headed lab tester." The idea is that anybody who takes a scientific approach to his hobby or profession and learns things that are unpopular or contrary to popular opinion is not human but some kind of unfit, unfeeling machine.

The implication is that 'Borgs are incapable of hearing the subtle differences that Golden Ears are so fond of alluding to. The reason why controlled listening tests tell us that "amps is amps" and wire is wire is that the test administrators and listening subjects are unfeeling and deaf. They are incapable of "getting it" and fond of faking test results.

Let's ignore for a moment that some of those loathed controlled listening tests were run by Golden Ears vainly attempting to confirm their silly notions and others used Believers (like Steve Zipser) as test subjects. Let's cut to the chase. The detractors are simply defending an indefensible position, slinging mud at the messenger. There is no experimental evidence that supports their position. Indeed, there are mountains of evidence to the contrary. So they are reduced to defaming anyone who actually has or might have a smoking gun.

A good friend of mine defines a Golden Ear as someone who can hear inaudible differences. In practice 'Borgs are superior listeners who have developed methods to help people avoid "hearing" inaudible differences. Differences that are a product of wishful thinking or listener bias are eliminated from Cyborg consideration.

There is no more feared citizen than the one who knows the emperor has no clothes and is unafraid to say so in public. I love Peter Aczel's *Planet of the Apes* metaphor. John Atkinson is Dr. Zaius, hiding the truth from his staff. In this scenario the Cyborg is to be truly feared because he is not afraid of the truth. He is not afraid to tell. He is analogous to the Charleton Heston character.

I hear-by, pun intended, unveil my 'Borg heritage. I do possess superior listening techniques. I refuse to knowingly let myself fall prey to listener bias. I refuse to spend money on things that fail to improve the sound quality of my system or increase my productivity. If I buy something just for the hell of it, I know the true cost and the true benefit. I will not pretend, even to myself, that I bought an amplifier for sound-quality benefits. I refuse to let inaudible sonic differences subtract from the quality of my sound system by stealing resources they do not deserve.

That said, I want to address a comment that John Atkinson made about my June 1998 *Stereo Review* article entitled "To Tweak or Not to Tweak." As you may recall, this was a single-blind, single-stimulus, controlled listening test, where a common CD player and loudspeakers were matched with an el cheapo "geek" stereo system, which was compared against a full-tweak system (outboard DAC, vacuum-tube preamplifier, high-end power amplifier, expensive interconnects, fancy speaker wire, plus careful system installation and wire dress). None of seven subjects was able to tell reliably which system was driving the loudspeakers when they didn't know in advance.

Of course, the Golden Ears made the usual complaints. The system was not "really" tweaked. The presentation method was low-resolution. (Of course, if I had used a switchbox they would have bitched about that.) I was biased. The subjects couldn't hear. The statistics were "inconclusive." And so on. Any experiment that fails to support the Golden-Ear agenda is automatically evil at the worst and "inconclusive" at the least.

Atkinson said the experiment "read" as bad science. He said there were a dozen variables being tested simultaneously, any of which could have made a large difference but just as easily could have cancelled each other out. According to John, it was a case of managing the experiment to get preordained results.

Boy, what a superior or incredibly lucky guy I must be. I just happened to stumble accidentally on exactly those dozen elements that all perfectly cancelled out? I don't think so. Perhaps my superior 'Borg hearing allowed me to select precisely those dozen variables that all perfectly cancelled one another. Only a Master Tweak could do that, I would think! With such abilities why would I want to hide the real differences from everybody?

Of course, this is the regular whining from the Golden Ear Society. This experiment was a real attempt to see if, as proponents claimed, a series of tweaks worked together in a synergistic fashion. It's really not my fault they don't. Information perceived as bad news is always unpopular, especially for wishful thinkers. Sorry, John. Life is tough.

On to greener pastures. At a Prairie State Audio Construction Society meeting last February, we dissected the Monster Cable M2.2s loudspeaker cable used for "To Tweak or Not to Tweak" to find out just what was in those mysterious networks at each end of the cable. One end has a can labeled Amplifier and the other is marked Speaker. There is a two-layer metal case at each end.

The outer layer comes off with threads. The inner layer has to be cut off. A Dremel saw took care of that. Underneath is a thick tough slug of potting compound. Ten minutes of deft work by Tom Perazella exposed the networks. At the amplifier end, the network consisted of nothing except the metal housing and potting compound. No capacitors, no inductors, no nothing, just wire.

The network at the speaker end contained a single component. Are you ready for this? The network was a single 100-ohm power resistor wired across the terminals. Yep. A 100-ohm resistor. The device will lower the impedance of your speaker system by a tiny fraction but it is electrically and sonically a no-show.

I suppose a network that does nothing and will have an unnoticeable effect should it burn out is preferable to one that screws something up! What a charade. No wonder the network is hidden in potting compound. I wouldn't want to show the real nature of my product to my customers, either, if I were selling this brand of snake oil.

So where is the industry heading now? Well, I think we are entering a new era where the topology of the audio system will eventually look like a computer. Right now we are a black box industry. Every time you get a new function, you buy another black box complete with chassis and power supply. Thanks to cost-effective VLSI integrated circuits, the metalwork and power supply are now the two most expensive parts of any component. They're too expensive with multichannel systems.

The fix is the telephone/computer style bus system and central processors. New functions are added with plug-in circuit cards and/or software. This topology is one reason telephone service has been so inexpensive for so long and computer prices have fallen so rapidly.

Audio is ready now. Furthermore, we have a new breed of enthusiast. The old guys, you and me, have a different set of baggage. The new guys learn about sound at the keyboard. They are interested in music but the computer is second nature to them. We will adapt and things will get even better even faster.

At the AES Convention I attended a workshop called "The Producer's View." Four recording engineers, George Massenburg and Alan Parsons among them, concluded that consumers are not going to spring extra bucks for a 96/24 DVD-Audio disc. Although they were all for a new expanded CD format, they realized that multichannel would be much easier to sell and most likely would be the next important medium. Amen to that! Two-channel is dead; long live the King.

So where does the hard-core enthusiast go from here? If new wires and new amplifiers and more vinyl are no longer performance upgrades, where do we go next? Here's my take. The best ways to improve your system, in this order:

1. Buy some new recordings. Your system is best improved from the outside in. Better recordings and better speakers make the biggest differences.

2. Reposition your speakers. Speaker position is the single largest performance factor in home reproduction. It's more important than the speakers you use.

3. Extend the bandwidth of your system. The separate subwoofer is necessary for optimal performance. And it is a DIY opportunity: you can still make a better one yourself than you can buy.

4. Get a new format. Multichannel is a major improvement. Dolby Digital is fantastic. Once you go to high-performance surround, you can never go back to two-channel stereo. Consider investigating the new virtual surround systems, which are really computer-based binaural played back through two speakers.

5. Get a remote control. Adjusting your system from your listening position makes a huge difference.

6. Get a picture. Consider a large-screen TV—you would be surprised at how much a picture improves imaging.

7. Make your own recordings. Digital recording and computer editing are relatively inexpensive these days, and you can do a remarkably good job in your den.

8. Renew your subscription to *Stereo Review*. It's still the best source of information available to the audio enthusiast. You get information expressed quickly and clearly, well in advance of any other source. Just because the news is not surrounded with bullshit doesn't mean it isn't useful. They also have editors with balls. Can you think of any other magazine, slick or underground, that would have published "To Tweak or Not to Tweak"? Technical Editor David Ranada is one of the sharpest tacks in the industry.

9. Everything else: Remember time is also a resource. Time spent auditioning wires can never be retrieved.

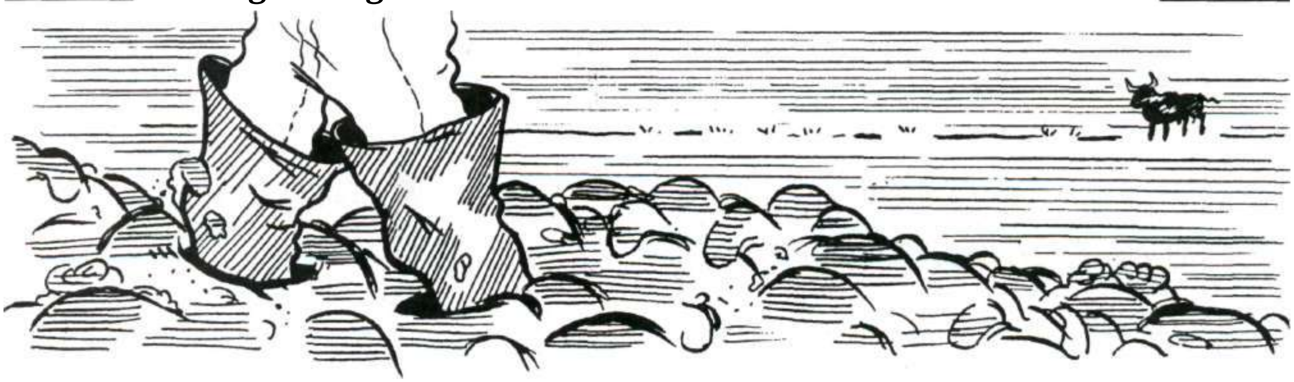
Editor's Note: When Tom Nousaine wrote the above, he didn't know yet that Hachette Filipacchi Magazines would combine Stereo Review *and* Video *into a single AV-oriented publication. I was going raise my editorial eyebrow in reaction to his unbridled adulation* of Stereo Review *(I share a great deal, but far from all, of his enthusiasm), but now it's all quite academic. At this point only the HFM high command knows—and maybe not even they—what might be the future direction of the new hybrid magazine. You can be sure, however, that it won't be anything HFM considers bad for business. (David Ranada is staying, as far as I know.)* •

A Note on Binaural CDs

At a time when the hot news (Sony, Dolby) is 5.1 surround sound decoded for headphone listening with directional and ambience cues supposedly intact, the obvious question that arises is—what about binaural recording? Isn't that the theoretically perfect headphone medium? I have received a few binaural CDs—recorded with microphones in the ears of a dummy head—the most interesting of which is Stravinsky's Le sacre du printemps *with Rachmaninoff's* Symphonic Dances *(Pasadena Symphony, Jorge Mester, conductor, Auracle/Newport Classic NCAU 10002). This is an excellent performance by moonlighting Southern California pros, but the sound fails to turn me into a born-again binaural religionist. Tremendous immediacy, yes; extreme detail, yes; directional information up the yingyang; but it's too much of a muchness, oppressive, disturbingly uncanny. It's like taking a bath in good wine, instead of sipping it from a glass. Not really the concert-hall experience.—Ed.*

Hip Boots

Wading through the Mire of Misinformation in the Audio Press



Editor's Note: *We have a new contributor girded for the fight against ignorance, antiscience, and tweako cultism. Glenn Strauss is not yet a burnout on the subject of cable idiocies (as I am and I think even David Rich is), so his first knightly foray here is into the wild world of wire warlocks.*

Synergistic Research: through a Forest, Darkly.

Synergistic Research can't see the forest for the trees. This company unapologetically touts the subjective, weighty importance of cables in home audio applications. That is not man-bites-dog news. But dog bites man on page 37 of their "Explorer's Guide to Synergistic Research Cables."

The topic is the controversial subject of cable burn-in. (Well, it is controversial only in the audio-pile press.) In attempting to explain this silly notion to the untutored, our guide uses the following:

"To better understand how cable burn-in effects [sic] the music you hear, it may be helpful to think of each frequency traveling through a conductor as a different trail or path through a forest. If you are traveling through the woods for the first time, and no trail exists, your going will be fraught with difficulty as you encounter rocks, thick bushes and dense forest. However, as you travel the same paths over and over, your going gets easier and easier. This is why cables seem to gain performance over time, and can actually lose performance or burn-in if they are not used in your system for long periods..."

I cannot begin to fathom what physical universe or laws could serve as the framework for such nonsense. Does the conductor undergo physical changes from an audio signal? Does the conductor even matter according to Maxwellian concepts? Do the audio frequencies "learn" the best route through the cables as a human being would in walking? Do cables forget what they have learned if not used regularly or if jostled?

This is just the kind of baloney that turns audio enthusiasts off and makes audio professionals shake their heads. This "guide" is fraught will other cable half-truths, unchallenged speculations, and sheer audio myths. It is also, like many high-end advertisements, chock full of spelling and grammatical errors.

One spelling error in particular may be revealing of a deeper meaning. On page 10, there is an attempt to educate the reader on "The Basics—Capacitance, Inductance, Resistance." Never mind that each section ends with a subjective interpretation of how these physical constraints change the "sound" of cables. But, according to Synergistic, capacitance is measured in "pico Farrats." Or is that ferrets? Or is it polecats? Or do I smell skunks?

—Glenn Strauss

[Ted Denney III is listed as CEO/Lead Designer of Synergistic Research. Isn't it possible that he cut English classes in school just as often as physics?—Ed.]

Robert Harley in *Fi* Magazine

Oh no! Please! Not Robert Harley again! But wait a minute, this is a little different. Harley is no longer with *Stereophile;* he is now Technical Editor of *Fi,* hired by founder/publisher/moneybags Jerry Gladstein (a former Harry Pearson disciple) in the belief that the Harley name would lend some scientific credibility to the magazine. Can you imagine? Of all the people they could have had, that's the one they went with. Of course, no one splits the difference between techie and tweaky talk as glibly as Bob, and that obviously filled the bill for *Fi.* Be that as it may, let me point out a classic Harley Howler to ruffle Jerry Gladstein this time, instead of Larry Archibald.

Harley's specialty is supposed to be the digital domain, right? Well, get a load of his editorial in the September 1998 issue, titled "Is Your CD Collection a Hidden Gold Mine?" He oohs and aahs over the revelatory improvement in CD sound quality ("dramatically increased transparency, space, treble smoothness, resolution, and harmonic accuracy") when the CD is played through an upsampling digital processor that samples at 96 kHz and/or 192 kHz instead of (as he understands it) the 44.1 kHz of a conventional CD player. He states that the higher

(continued on page 62)

The Audio Cynic

By Glenn O. Strauss

By Glenn O. Strauss

Editor's Note: *As a parody name for our journal, "The Audio Cynic" was first publicized in 1977 by a disgruntled amplifier manufacturer whose product had gotten a bad review in one of our earliest issues. It happens to be a timeworn witticism, but when Glenn Strauss wanted it as his handle for his new column, my reaction was "why not?".*

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I was a part-time, weekends-only salesman in a high-end audio salon during the high end's heyday in the late '70s and early '80s. A computer engineer by profession, and later technical editor for an audio "underground" quarterly, I found it necessary to enter the inner sanctum of audio's holiest shrines in order to afford the ever-increasing price of admission. The tale that follows, and others I might get around to, are meant irritate, educate, placate, and elucidate—and maybe even provide a laugh or two. But they are 100% true.

The Audiophile from Hell

Every high-end audio shop has experienced the intense fear and loathing that follow the peculiarly irritating "customer" who really isn't one at all. Here is the story of one particularly obnoxious fellow. Before the whole sick thing was ended, his behavior moved from odd to rude to sociopathic.

It all started innocently and typically enough. A young professional showed up in the store one day, bedecked in the uniform of the Southern intelligentsia—golf shirt, baggy chinos, deck shoes. He seemed interested in a number of the high-end lines we carried, particularly some of the electronics. And he seemed sincerely interested in a purchase at some point.

The first indication that something was not right in the seller-buyer relationship was the way the guy asked questions. Most of the questions were quite reasonable in that they were directed towards a better understanding of our products, why we thought they outperformed other products, and what kind of support both we and the manufacturer offered to our customers. But he had the irritating habit of asking another question well before the first one was answered. At first we just shrugged it off as a personal mannerism but later realized he wasn't really interested in what we had to say. Instead, he was trying to put in enough time at our dealership to demand some loaner service.

The loaner process is unique to the high end in the consumer electronics business, The seller maintains an inventory of expensive equipment that can be borrowed by a customer and used for an often surprising amount of time at no cost, with absolutely no commitment to buy. It is part of the dealer service and is partly covered by the dealer markup, All risk is to the dealer, and we often had equipment returned damaged, cosmetically or electrically. We always absorbed the cost, even when someone's kid poured a soft drink into a $3000 Threshold amp. Did we ever have a customer offer to pay, even when many homeowner's policies would have covered the damage? Not that I recall.

Back to our fledgling audiophile. Over the period of a year, this guy borrowed preamps, amplifiers, and any number of large panel loudspeakers, all of which we brought to his home, set up, picked up, and returned to the shop. Often these were pieces so expensive that we could not afford duplicates in the store. That meant that we sometimes went a week without having a product on display. This was not uncommon practice, and one we gladly performed for our buying customers,

After a year, it became clear that something was amiss. We tried to pin the guy down, but he deflected most of our questions by saying this was going to be a big purchase and he wanted to make sure he was getting products that pleased him. Hard to argue with that logic. That was until we learned he was buying the same equipment at discount, from a dealer about 200 miles away! For obvious reasons, our attitudes toward this guy changed immediately, once we learned he was an audiophile vampire, living off the lifeblood of others. His loaner privileges were revoked, although he was allowed to come into the store to "rap" (hey, this was the '70s, after all).

We maintained one of the largest assortments of electrostatic loudspeakers in the Southeast. For all their flaws, we were convinced that they were the best things going in terms of transparency and transient response. Our vampire was particularly argumentative that some cone loudspeakers could do the job. In a moment of guard dropping, he spilled the beans—he had purchased a pair of AR Model 9's. We knew there was no AR dealer in town, so they had to have been bought mail order. That was the last straw, or was it?

This worm set a new standard of audiophile effrontery about seven months later when he showed up at the store one Saturday, wife in tow. (I have tried to keep the wife out of the picture because she was never really a player, except for the common spousal practice of viewing us as children of Satan, hell-bent on destroying architectural progress in her home, and the only thing standing between her children and an Ivy League education.) When we asked him what he wanted, he opened a little cardboard box in which a dome tweeter nestled. We looked at the tweeter, and it wasn't one we recognized, so we looked questioningly at the guy. "It's from my AR Model 9—it's defective." Never mind that we had never seen a defective tweeter, unless you call a melted voice coil a factory defect. But that's another story.

When asked what he wanted us to do, he indicated he wanted us to replace it. When we pointed out that we didn't sell him the speakers in the first place, his reasoning was that since we were an audio shop and repaired equipment, we could fix things for him by acting in his behalf with AR. We suggested he get the selling dealer to assist him, but his response was that "that would be a hassle." Apparently any hassle we might experience was not relevant. We finally said we would be willing to get him a new tweeter for its retail cost plus shipping.

The guy went berserk. From somewhere in the vast nothingness of his mind he started to dredge up increasingly bizarre arguments about why we should be acting in his behalf. He was "a friend of the store," an odd claim since he had never bought a damn thing and had wasted many hours of our time. (Remember, it is Saturday and we are busy, and his voice is starting to exceed conversational levels.) He then formulated his ultimate absurdity, the social contract argument: because we were an audio store, we had a moral imperative to assist him. I kid you not.

We asked him to leave the store. At this point his wife got into it. (Who says opposites attract? Apparently these two were cut from the same philosophical cloth.) She opined that our store was a public place, and that they could not be asked to leave. We suggested that unless he, she, and the abused tweeter did not leave immediately, we would call the police. They left, as had half the people in the store by that time. Who knows how many potential sales we lost?

We estimated that approximately 15% of the people who came to our store saw nothing ethically wrong in stealing time and "sweat equity" from us. I have always felt that this is the sort of thing that forced more and more tweako product sales, since a $5000 speaker cable has at least 50 points profit, is a carryout, and is unlikely to be damaged.

Oh yeah, the audio vampire called back several months later after everything had cooled down—he wanted to borrow some more product. It was then I decided that the chances of my investing any money in a high-end audio store were about as good as his chances of getting any more loaners. •

Hip Boots *(continued from page 59)*
sampling rate permits a gentler filter at a higher frequency with a less steep rolloff, hence "far less time smear."

The mind reels. Is it possible that Harley doesn't know that for the past decade he has been listening to ordinary CD players with 4-times and 8-times interpolating digital filters? That's upsampling (oversampling in vulgar parlance) to 176.4 kHz and 352.8 kHz, no different from what he admires so much in the $5750 processor his comments are based on. Well, it appears the "digital lad" (his pet name at *Stereophile)* doesn't know that. Maybe the word "upsampling" is new to him, so he thinks the technology is new. He writes that "upsampling converters may become commonplace in high-end playback systems. I can't wait."

My take on this is that John Atkinson at *Stereophile* still exerted some sort of minimal control over Harley's technical fantasy life, but at *Fi* the lad is totally out of control because there is no one to question him. Jerry Gladstein, who sold you this bill of goods?

I think what Bob needs now is a nice $120 Sony Discman with that incredibly sophisticated, futuristic upsampling/oversampling/interpolation capability. The harmonic accuracy will astound him.

—Peter Aczel

Peter van Willenswaard in *Stereophile*

One of the purposes of engineering conventions like those of the AES is to bring new untested ideas out in the open. During the question period at the end of each paper presented, the merits of the ideas are openly discussed. AES audits only the 500-word precis, not the full paper; thus some bad ideas make it through, but the identification of such problems is part of the purpose of these conferences.

A recent example of the working of this process is a paper presented by Johannes M. Didden on September 29, 1997, at the 103rd Convention of the AES in New York. Titled "Novel Feedback Topology Obviates the Need for High Loop Gain," the paper was distributed as Preprint 4597. The idea Didden presents is that by adding another loop to the standard feedback topology, high loop gain is no longer required in a feedback system. Unfortunately some pole-zero cancellation in the mathematics causes a singularity to be hidden. In fact, as pointed out during the q-and-a session, the extra loop introduces positive feedback with a loop gain of one. In the actual realization, compensation methods prevent the amplifier from oscillating, but as another questioner pointed out they also prevent the amplifier from having any performance advantage over normal negative feedback. As a final blow, it was pointed out that even if stable during normal operation, the amplifier could latch to a supply rail if clipped. Note that nothing is wrong here—this is how the academic process works, and Mr. Didden is to be congratulated for having the courage to bring his ideas before his peers for testing.

On the other hand, *Stereophile*—in their December 1997 issue, pages 35 and 37, "Industry Update" section, under Peter van Willenswaard's byline—reported this development in a very different manner than the clinical Didden paper, which makes no sonic claims for the system. "…I have become very suspicious of (overall) feedback... feedback tends to kill the magic in the music...To me Didden's proposal is a major step forward in feedback theory." It looks like *Stereophile* has once again attempted to canonize an idea as having sonic benefits even if the idea itself is all wet. Note that the AES convention took place in late September, so *Stereophile* should have had enough time to kill, or at least modify, an article that hit the street two months later, but they chose not to do so. What we have here is yet another example of *Stereophile* not taking the time to do the required fact checking before they rush to publish.

—David Rich

Capsule CD Reviews

By Peter Aczel
Editor and Publisher

(Note that the year in parentheses after the CD number is the year of recording, not the year of release.)

ABC Classics

The initials stand for Australian Broadcasting Corporation; the label has been around for a number of years, but its introduction to this country is quite recent. Australia is by far the most remote civilized place from our vantage point and remains a source of wonderful musical surprises because that's not where we generally look for them.

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***"Rita Hunter: Ritorna Vincitor!"** Arias from Verdi, Ponchielli, Puccini, Beethoven, and Mozart operas. Rita Hunter, soprano; Tasmanian Symphony Orchestra, Dobbs Franks, conductor. 8.7000 10 (1989).*

I chose to review this relatively ancient recording out of a good many sent to me by the distributor because of the astonishing voice of Rita Hunter. I have no idea whether she still sounds as good today as in 1989, but on this CD she is fantastic. Her voice is big, warm, utterly secure, free and unstrained on top—I can't fault it. Her musicianship is also of the first order. I played her *Vissi d'arte* side by side with Callas's from the 1953 *Tosca* recording on EMI, and I swear I can't make up my mind which is better. (Callas is more dramatic, more riveting; Hunter is more controlled and more beautiful in sound.) The Tasmanian orchestra plays very well (of course, these accompaniments are not terribly demanding), and the digital recording by the ABC leaves nothing to be desired. Throw a shrimp on the barbie, mate, and listen to this one.

Celestial Harmonies

Most of the releases on this label have only niche-market appeal (music of Islam, didjeridoo, gamelan, that sort of thing), but now and then they come up with a winner for us mundane classical-music lovers stuck in the three-B's rut. (Hey, it's a rut even if it's Bartók/Berg/Boulez.)

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***Heinrich Schütz: Der** Schwanengesang, SWV 482-493. The Song Company, Roland Peelman, artistic director. 13139-2 (1996).*

This is as different from the currently trendy New-Age-flavored "chant" releases, which bore me to tears, as prime filet mignon is from a Big Mac. Schütz was born 100 years before Bach, and this superb cycle of motets (for eight voices in two antiphonal choirs with organ continuo) is all the evidence needed to prove he was a very great composer. The style is a blend of homophony and counterpoint. The performance is by a truly superior ensemble of Australian singers. The recording, a very clean job, was made at the Sydney Opera Concert Hall, which has lovely acoustics. Excellent booklet, too. Recommended.

Chandos

For no particular reason, by mere happenstance, this fine English label has been neglected by this journal. Here is one of their recent releases.

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***Frédéric Chopin:** 24 Preludes, Op. 28; Prelude, Op. 45; Andante spianato et Grande polonaise brillante, Op. 22; Polonaise-fantaisie, Op. 61. Louis Lortie, piano. Chan 9597 (1997).*

"We have been, let us say, to hear the latest Pole/ Transmit the Preludes, through his hair and fingertips."—T S. Eliot. Actually, Louis Lortie is Canadian, not Polish, and has a short haircut; even so he "transmits" the Preludes just as poetically. Indeed, poetry is everything in the performance of these extremely short pieces, and Lortie is equal to the interpretive challenge. The other opuses are also performed with considerable technique and musicality, but the supply of great Chopin playing available on CD today makes it very difficult for any pianist to make an important additional statement—and that is neither expected nor accomplished here. The Chandos 20-bit recording yields a truly superior piano sound.

Chesky

Known for perfectionism in sound, occasionally rendered questionable by genuflections to the tweako element, this label addresses both the classical and non-classical markets and features newly recorded as well as remastered releases. Their best work is impeccable.

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***David Chesky:** Three Psalms for String Orchestra. Deutsches Filmorchester Babelsberg, Stephen Somary, conductor. CD163 (1997).*

On the box it says "High Resolution Technology" and "Recorded at 96/24." That refers to the digital master; the CD is still just a CD, and that means 44.1/16. But what a wonderful-sounding CD! The strings have just the right weight on the bottom and are never, never harsh on top, even when the music gets loud. Why can't the major-label strings sound like this? As for the music, it is eclectic, fairly melodious, totally accessible, maybe a bit monotonous, but nice listening overall. Those who like the Samuel Barber "Adagio for Strings" *(Platoon* soundtrack) will probably enjoy it. Oh, yes, the composer owns Chesky Records, with his brother.

Delos

If I were asked to award Olympic gold, silver, and bronze medals to present-day recording engineers, it would be a tough call, but I think the gold would go to John Eargle of Delos. He, and he alone, is without a techie/politico agenda regarding microphones, microphone placement, electronics, etc. He doesn't want to prove anything to other engineers or to audiophiles; he just wants great sound and goes after it any way he can. His best work defines the state of the art, and very little of his work strays from his best.

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***Engineer's Choice II:** "Top recording engineer John Eargle picks his favorite demo tracks." DE 3512 (1996 and earlier).*

In 1991 Delos released John Eargle's original *Engineer's Choice,* with the same subtitle. This sequel also has 22 tracks, just a few of which overlap with the older CD. Most of the tracks are new, however, and thus more representative of John's latest and greatest efforts. Put this CD in your pocket when you go a-hi-fi-ing; it's comprehensive and portable reference material second to none. Check out the 20 Hz pedal notes in the Messiaen organ excerpt and the incredible orchestral definition in the Shostakovich 8th Symphony excerpt. The John Eargle signature qualities are the total lack of strain, regardless of dynamics, and the wide-open, panoramic soundstage. (Yes, tweaks, soundstaging comes from the recording, not from your overpriced amplifier.)

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***DVD Spectacular:** Tchaikovsky's "1812 Overture" (Litton/Dallas) and other demo tracks, audio tests, video tests, etc., for AV surround-sound systems. DV 7001 (1997).*

This is not a CD but a DVD, which arguably belongs in a different review column. (We are planning to have one in all future issues.) The reason for its inclusion here is that it features a Dolby Digital (5.1 discrete surround) version of the same performance of the "1812 Overture" whose sound quality I praised so enthusiastically in the last issue. Well, if the 5.1 system is properly set up and trimmed in, the sound is even more impressive here. This is probably the most realistic orchestral/choral sound you can have in a well-equipped domestic listening room. I am almost ready to admit that the best two-channel stereo leaves something to be desired. (Of course, not all Dolby Digital sound is authored by John Eargle.) The test signals on the disc are useful, but that's another discussion altogether. And don't expect any video footage of the orchestra. This is one of those newfangled audio DVDs.

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***DVD Music Breakthrough:** 16 tracks from the Delos catalog (Litton/Dallas, DePreist/Oregon, Macal/New Jersey, etc.) in Dolby Digital and stereo versions. DV 7002 (1998).*

***DVD Space Spectacular: Strauss**'s Also sprach Zarathustra and **Hoist's** The Planets in Dolby Digital and stereo versions. Dallas Symphony Orchestra, Andrew Litton, conductor. DV 7003 (1998).*

These two audio DVDs, sequels to Delos's original "Spectacular," came in after the above review was written. They are not an improvement in surround sound over DV7001; I was actually less impressed—but maybe only because of the somewhat less AC-3-genic program material. There is some good music here, and the Strauss/Hoist performances are also available on conventional CDs *(DE 3225, 2 CDs, 1997).* Unfortunately, Litton's *Zarathustra* is quite pedestrian; he does not seem to be comfortable in the great Austro-German tradition, making little effort to phrase the magnificent string passages passionately. The Hoist warhorse requires more technique than *Innigkeit* and is much better performed. John Eargle's recording of *The Planets* is the only one known to me that never, ever, turns harsh and unpleasant in the climaxes; it is worth having for that reason alone, on audio DVD or CD.

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***Hector Berlioz:** Symphonie fantastique, Op. 14; Romeo et Juliette, Op. 17, Scène d'amour. New Jersey Symphony Orchestra, Zdenek Macal, conductor. DE 3229 (1997).*

Audiophiles will want this CD because it is arguably John Eargle's best work to date—and that means the best there is. It is the first orchestral recording made in the recently completed New Jersey Performing Arts Center in Newark, obviously a great venue. Music lovers, on the other hand, will be able to cite much better performances, especially of the magnificent "Love Scene," which appears to be underrehearsed by this part-time orchestra of top musicians from the New York area. The *Fantastique* receives a respectable performance; Macal is after all a conductor of some stature; but the obvious star here is the sound—a case of the best getting better. A DVD version is in the pipeline.

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Erich Wofgang Korngold: *The Sea Hawk; Symphony in F-sharp, Op. 40. The Oregon Symphony, James DePreist, conductor. DE 3234 (1997).*

Lots of interesting angles here. Korngold was 6 years younger than Prokofiev and 9 years older than Shostakovich, and his idiom fits right in there, in terms of reined-in modernity and high accessibility, with a bit of Mahler nostalgia thrown in. Not that he is quite as good a composer as any of those three but he is clearly a master in his somewhat shallower Hollywoodsy way, a fabulous orchestrator, and never boring. The symphony, which took him three decades to complete, is a massive 54-minute work for huge orchestra. The Oregonians under DePreist play it so well that you could have fooled me if you told me I was listening to one of the biggies. And that's not all. In a new venue for his VR^2 recording technique, John Eargle proves once again that he is a little better than the best of the rest. This is a demo/test disc for big systems if there ever was one—and the music actually bears repetition!

Denon

I was wrong in the last issue. There are plenty of new releases on this label.

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Franz Joseph Haydn: *6 "Erdödy" Quartets, Op. 76, Nos. 1-6. Kuijken String Quartet: Sigiswald Kuijken & Francois Fernandez, violins; Marleen Thiers, viola; Wieland Kuijken, cello. CO-18045/46(2 CDs, 1995-96).*

The Kuijkens' period practice doesn't set my teeth on edge nearly as much in Haydn as in Mozart (see Issue No. 22, p. 54), but I can't say this is my favorite way to hear these superb quartets, which are among Haydn's best. I admire the sure-handed authority, polished ensemble playing, and unshakable musicality of these scholarly artists; I realize that the style is *echt* Haydn; but I want I little more vibrato, a little more anachronistic expressiveness, a little less four-square simplicity in my 1797 Viennese music. I know: my taste has been corrupted by the Romantic performance style. As for audio quality, the German-Japanese recording team did a fabulous job in three different locations, achieving total transparency and unstrained dynamics. The slightly nasal string tone is true to life, not an artifact.

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Leos Janácek: *Msa Glagolskaja (Glagolitic Mass); Sinfonietta. Julia Varady, soprano; Stella Doufexis, mezzo-soprano; Valentin Prolat, tenor; Peter Rose, bass; Rundfunkchor Berlin; Arvid Gast, organ; Deutsches Symphonie Orchester Berlin, Eliahu Inbal, conductor. CO-18049 (1995).*

Two of the indisputable masterpieces of the century, in very intense, committed performances. The Berlin orchestra plays with considerable virtuosity, and the singers are excellent, although I can't guarantee their pronunciation of Old Church Slavonic. Maybe a Janácek specialist can find fault with these performances, but I can't. The audio quality is right up there with Nippon Columbia's best, which is second to none. It all adds up to a very satisfying musical experience.

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Richard Strauss: *Also sprach Zarathustra, Op. 30; Till Eulenspiegels lustige Streiche, Op. 28; Macbeth, Op. 23. Orchestre de la Suisse Romande, Eliahu Inbal, conductor. CO-18067 (1995-96).*

I obsessively listen to all—well, nearly all—new recordings of *Also sprach Zarathustra*, the audiophile's benchmark piece. I regret to report that this one doesn't particularly excel in any area, be it interpretation, orchestral playing, or audio quality. That doesn't make it bad, just routine. Maybe Inbal had to deliver this package quickly, without sufficient rehearsals. *Till* and *Macbeth* are roughly on the same level. The latter, one of the very early and lesser tone poems of Strauss, is seldom recorded, but Schwarz/Seattle (1990) on Delos is better. Can't win them all.

Deutsche Grammophon

I continue to like the sound of DGG's 4D Audio Recording. Why couldn't it have happened sooner? Think of all those Karajan, Bernstein, etc., performances!

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Ludwig van Beethoven: *The String Quartets. Emerson String Quartet: Eugene Drucker and Philip Setzer, violins; Lawrence Dutton, viola; David Finckel, cello. 447 075-2 (7 CDs, 1994-95).*

This is the set I previewed (having auditioned producer's DATs of three of the quartets) in Issue No. 23. I wrote: "this will be the set to own, above all others." Now that all 16 quartets are available on these 7 CDs, I see no reason to change that opinion. At the same time, I am aware that not all critics agree with the Emerson's Beethoven style. No one denies their amazing virtuosity and perfection of ensemble, but some feel that their interpretations are too "modern," hard, aggressive, literal, unrelaxed, unlyrical, *ungemütlich*, or whatever. I, on the other hand, believe that what we have here is as close to the music Beethoven heard in his head as we are ever likely to hear. (He still had his hearing when he composed Op. 18, Nos. 1-6, so I am basically talking about the ten quartets that followed.) Max Wilcox, listed as both recording producer and balance engineer for the set, exerted more than the usual producer/engineer's influence on the recordings and was one of those who encouraged these four great string players to depart from their accustomed tempi and follow closely Beethoven's metronome markings. Eugene Drucker, who alternates between first and second violin in the Emerson's performances, explains in the program notes that the pet theory of metronome error in the early 1800s, as advanced by some writers, simply does not hold water. I find the tempi in these recordings to be exactly to my liking, thrilling in the fast movements and exquisitely flowing in the slow ones. Of course, no quartet can get through the entire Presto (scherzo) movement of Op. 131, nor the concluding Allegro, at the Emerson's tempo and with the Emerson's attack without making a single mistake, but the editor (Max) can make it happen. And that's just one example. No, these aren't documentaries of actual performances but completely idealized renderings of the music. To me they appear little short of miraculous, and I do not miss the verismo of a "live" event. The audio quality, as I already reported last time, is also quite sensational—as natural, transparent, and detailed as I have ever heard in a quartet recording. In sum, a landmark set and a joy forever.

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Johannes Brahms: *Concerto for violin and Orchestra in D Major, Op. 77.* ***Robert Schumann:*** *Fantasy for Violin and Orchestra in C Major, Op. 131. Anne-Sophie Mutter, violin; New York Philharmonic, Kurt Masur, conductor. 457 075-2 (1997).*

The Brahms is the featured work here; the Schumann is a 13-minute filler. The Mutter/Masur performance of the concerto is almost incredibly good; both violinist and conductor are caught here on their best day in a live performance before an audience. Mutter is virtuosic, expressive, and romantic to the nth degree; the Philharmonic plays as if the year were 1936; and Masur makes it all happen. The recorded sound is a Martin Fouqué triumph over the wretched acoustics of Avery Fisher Hall—remarkably rich and beautiful. But then I took out the 1955 Heifetz/Reiner/Chicago recording in the remastered Living Stereo edition and quickly realized that Anne-Sophie Mutter is to Jascha Heifetz as Drew Bledsoe is to Joe Montana. Not quite there yet.

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Frederic *Chopin: Fantaisie in F Minor, Op. 49; Piano Sonata No. 3 in B Minor, Op. 58; 3 waltzes; 3 études; et al. Mikhail Pletnev, piano. 453 456-2 (1996).*

Pletnev is barely forty and already an internationally celebrated conductor, as well as a pianist. His Chopin is far from straightforward, quite mannered in fact (especially in the great Fantaisie), but he projects a remarkable musical personality in every phrase, and in the end one is totally captivated by the beauty of his playing and dazzled by his virtuosity. The recorded piano sound is excellent.

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George *Frideric Handel: Music for the Royal Fireworks, HWV 351; Concerto in F Major, HWV 331/316; Concerto in D Major, HWV 335a; Passacaille, Gigue and Menuet in G Major; Occasional Suite in D Major. The English Concert, Trevor Pinnock, harpsichord/musical director. Archiv 453 451-2 (1996)*

Those who are familiar with my usual sour comments on period practice will be surprised. This is period practice with a vengeance—18th century instruments, A tuned to 415 Hz, unequal temperament tuning, etc.—and it's wonderful! The famous "Royal Fireworks" piece is played in the original 1749 version with 24 oboes, 12 bassoons, contrabassoon, 9 horns, 9 trumpets, 3 timpani, and 3 side drums. Put that in your CD player and turn the volume up! And that's not all. Part of the F major concerto is a reworking of the earlier *Water Music*, and there are quotations, adaptations, and cross-references across the board in the other pieces. Handel loved to quote himself. Pinnock's orchestra plays this Handel fest with tremendous panache; the "rhythm and pace" here come from the musicians, dear tweaks, not the speaker cable. Archiv's all-German recording team did a great job with the sound in an English hall. Highly recommended.

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W.A. Mozart: *Opera Arias ("Kathleen Battle Sings Mozart"). Kathleen Battle, soprano; Metropolitan Opera Orchestra, James Levine, conductor. 439 949-2 (1993).*

DGG sat on this for four years before releasing it, probably because of the embarrassment of Battle's expulsion from the Met. That doesn't make her a less good singer. She can do just about anything with that not very big but very pretty and beautifully trained voice, and she has the ear of a musician. Her high tessitura occasionally strays from her absolute best, and her lower tones are not the warmest possible, but she is still a very distinguished soprano. As a singing actress she could be quite a bit better; her Countess, Susanna, Cherubino, Zerlina, and Pamina all sound like the same lovely, polished singer, with little or no character differentiation. There are 13 arias from 7 Mozart operas on this CD, all of it as good as it gets. The Met

An Object Lesson in Creeping Subjectivity
(How We Unfairly Suspected Deutsche Grammophon of Doctoring the Sound)

As our readers know, ***The Audio Critic*** is pretty doctrinaire when it comes to listening comparisons. Double blind, levels matched within ±0.1 dB, a meaningful number of trials, no excuses. This little cautionary tale is about what happens when those rules are not followed. The kicker here is that the offenders were none other than your Editor and Dr. David Rich, which is the audio equivalent of catching two bishops in the whorehouse.

I said no excuses. Well, we did have an excuse for letting our guard down a little bit. We weren't comparing audio equipment; we were comparing recorded music. It's a terrible excuse because the discipline should be equally rigorous in either case.

What happened was that I fell in love with some DAT "previews" of the Emerson String Quartet's new set of the Beethoven quartets. Max Wilcox was in the process of producing these new recordings for Deutsche Grammophon and had been kind enough to lend me the DATs in order to give me a foretaste of a project of which he was enormously proud. Rightly so because the recordings brought an unprecedented level of virtuoso string playing to these much-recorded masterpieces—absolutely breathtaking attack, synchronicity, intonation, and clarity—and at the same time adhered religiously to the tempi specified by Beethoven's metronome markings. As for the audio quality, I heard the purest, most natural, most believable string sound out of my reference system, unsurpassed by any quartet recording known to me. This was going to be one of those alone-on-a-desert-island sets: the world's greatest music (isn't it?) in the world's greatest performances (in my instant opinion, anyway) and the world's best sound (at least to my ear). David Rich, who usually tries to one-up me with something "better" that he knows and I don't, actually agreed with me (sort of, as much as he ever does).

Then, after a somewhat longer delay than expected, the seven-CD set was released by DGG. David rushed out and bought it at once, before I could get a review sample. He brought it over; I inserted into the CD player a quartet we had already heard on DAT; and we listened. Almost at once, as if we had never sloughed off the conspiracy theories of our early tweako years, there came the exclamation: "The bastards changed the sound!" Let's face it, DGG is not exactly a hero to old-time audio purists; the marketers' allegiance to Max Wilcox, who was the producer/engineer on the Emerson players' insistence, did not appear inviolable; and the sound did seem brighter, more aggressive, less refined than what we remembered. We were outraged. The desert-island treasure had been vandalized.

I then played Max's DAT again to check the same passages, and we concluded that maybe the CD version wasn't all that terrible but still quite audibly different, probably as a result of a little EQ snuck in there in anticipation of peasant tastes. Oh yes, we did match the levels of the CD player and the DAT deck—by ear. After all, as string quartet aficionados, we knew how to detect gross differences without endless fussing. "Just listen," we kept saying in the best Bob Harley tradition, totally out of touch with our knowledge base.

And that's not all. I called up Max Wilcox, knowing that he listens to his master tapes a lot more regularly than to the CDs, and informed him of DGG's "treachery." He was both upset and incredulous. It is utterly impossible, he told me, that anyone should have touched the sound he had approved as producer. That's not the way the system works, he said. At the same time, I sensed that I had planted a seed of doubt in his mind. Holy PolyGram, what if Peter is right...

I received a call from Max the very next day. He had carefully compared the CDs with his DAT copies of the masters, and there was nothing wrong; the two sounds were identical. What kind of tweako cultist have I become, anyway? (That's not what he said, but the implication was there. I had obviously caused him some grief.) For the first time, I began to think that maybe David and I had been careless and jumped to a false conclusion.

I then set up a bulletproof comparison. Instead of switching between the line outputs of the CD player and of the DAT deck, with the attendant level-matching pitfalls (not to mention two entirely different D/A converters and two different analog output stages), I used the digital output of each, plugged into an outboard D/A processor with several digital inputs. That way the level matching was automatic, as long as 0 dB was set identically in both digital sources, and I just had to switch between the two through the same DAC and the same analog circuitry. (Remember, what Max had recorded was simply a bitstream, which the CD people had either tampered with or not.) Lo and behold, the two sounds were now indistinguishable. Max's sound was clearly inviolate. Red faces in our listening room, with egg on top.

Obviously the flaw in the original listening comparison was that the CD playback level was marginally higher than the DAT level, maybe by as little as 1 dB or even just 0.5 dB. You can't match it much better than that by ear, and such a small difference isn't necessarily perceived as louder/softer but as a quality difference. It is also possible that we were even sloppier, maybe by several dB. The sound of string music, in particular, can vary along the sweet-to-edgy axis depending on loudness. In a given room, through a given audio system, there is really only one level that sounds utterly natural. We screwed up, bigtime.

Do I have to spell out the moral? There is no meaningful listening comparison at even slightly mismatched levels. Even in a double-blind test, you first have to match the levels, otherwise everything will sound different and the test will be worthless. It's Rule No. 1. We knew that all along but we forgot. And you know Rule No. 2: never forget Rule No. 1.

—Ed.

orchestra is wonderful and so is Levine. The sound, recorded in New York's Abyssinian Baptist Church, leaves nothing to be desired. A success, all in all.

dmp

I like Tom Jung's work so much, just for its sheer sound quality, that I try to force myself to like the music he records but I succeed only occasionally. Yes, I rather like the music on the following CD.

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Bob Mintzer Big Band: *"Latin from Manhattan." Bob Mintzer, saxophone; 16 others (saxophones, trumpets, trombones, piano, bass, drums, percussion). CD-523 (1998).*

Ten tracks, most of them Latin-flavored, all of them played with considerable verve by these fine big-band musicians. (Try track 8 for some truly virtuosic horn solos.) The sound is quintessential Tom Jung: fantastic in-your-face presence with hard left/center/right localization. Of its kind, it's state-of-the-art; there's nothing better. Good speaker test.

Dorian

The superior audio quality of Dorian releases can hold my attention just so long; then the music has to take over, and their most recent repertoire just doesn't do it for me—with a few exceptions.

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"For Your Ears Only." *Music from James Bond movies, etc. Proteus 7 (seven-piece band). xCD-90258 (1997).*

Brian Levine, Dorian's rather elitist A-and-R man, must have been holding his nose when they made this audio-goon spectacular. It features a 24-bit digital recording technology developed by Craig Dory and yclept xCD. Imagine the dmp type of sound that jumps out at you, but with a rounder, acoustically more graphic, more suavely delineated quality, since the venue was the great Troy hall. It's a sound that will indeed impress you and go into your demo collection if you don't cringe when you hear the da-da-dum-*dee*-dah James Bond signature theme on steroids (not to mention the jokey sound effects). What a classical record label won't do for a little extra income...

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Joseph Haydn: *"The Hidden Haydn." Symphony No. 12 in E Major; Symphony No. 64 in A Major ("Tempora Mutantur"); Symphony No. 44 in E Minor. Apollo Ensemble, John Hsu, conductor. DOR-90226 (1995).*

I should have reviewed this extraordinary CD two issues ago, when it was new, but didn't. There's nothing like it among the latest Dorian releases. When I want to demonstrate the highest degree of sonic credibility, of you-are-there realism achievable in a two-channel recording, this is the CD I take off the shelf. The palpable presence of John Hsu's 14-piece period-instrument chamber orchestra between the two stereo speakers is unequaled in my experience. Brian C. Peters, who is no longer with Dorian, was producer, engineer, and editor of the recording. He moved in a little closer to the orchestra in the Troy hall than is the general Dorian practice, and the result is magic, at least on this kind of music. Just listen to those strings! That the music is great ("hidden Haydn" because these symphonies remained unpublished during most of Haydn's life) and the performances outstanding is a mere bonus to the flabbergasted audiophile.

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Heitor Villa-Lobos: *String Quartet No. 7 (1942); String Quartet No. 15 (1954). Cuarteto Latinoamericano: Saúl Bitrán &Arón Bitrán, violins; Javier Montiel, viola; Alvaro Bitrán, cello. DOR-90246 (1996).*

A relatively recent Dorian release, this is wonderful music, beautifully played and recorded. I find the earlier quartet more immediately captivating, but both are outstanding examples of comprehensible 20th-century music. The three Bitrán brothers and their violist partner have the idiom down pat, and the Mexico City recording sounds every bit as good as stateside Dorian. Highly recommended.

EMI

This is another great label that has had insufficient coverage in our pages. Here are two fairly recent releases of more than ordinary interest.

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Alexander von Zemlinsky: *Die Seejungfrau (1902-03); Sinfonietta, Op. 23 (1934). Gürzenich Orchestra/Cologne Philharmonic, James Conlon, conductor. 7243 5 55515-2 (1995).*

Alexander von Zemlinsky: *Der Zwerg. Soile Isokoski, the Infanta; David Kuebler, the Dwarf; Iride Martinez, Ghita; Andrew Collis, the Chamberlain; Frankfurter Kantorei, Gürzenich Orchestra/Cologne Philharmonic, James Conlon, conductor. 7243 5 56208-2 (1996).*

Seven years younger than Richard Strauss, Zemlinsky (1871-1942) was Arnold Schönberg's brother-in-law and sounds a little bit like Strauss edited by Bernard Herrmann (I am only half kidding). It's a postromantic sound, very beautifully orchestrated, maybe a little too slick, too much like movie music—but so much better than a lot of stuff in the permanent repertory. What it needs is some good PR, and James Conlon is just the man for that. He is the principal conductor of the Paris Opera and a great admirer of Zemlinsky's music. The Cologne orchestra, of which he is also chief conductor, plays all out for him, both in the purely orchestral pieces on the single CD and in the two-CD opera set, which features the excellent tenor David Kuebler in a very difficult role. *Der Zwerg* (The Dwarf) is considered by many critics to be the composer's masterpiece, a one-act "sicko-drama" based on an Oscar Wilde play, just like (well, not unlike) Strauss's Salome. If you're into the Mahler/Strauss/early-Schonberg bag, you'll almost surely like this music, as you will the German-engineered sound. Good show.

London

Other major labels have upgraded their sound over the past few years (DGG and RCA come to mind), but English Decca has not, perhaps because they feel, not without some justification, that their sound was very good to begin with. Even so, their recent releases could do with a bit more precise spatial delineation and a slightly less hot top end to be competitive with the absolute best of today's recordings.

Renée Fleming: The Beautiful Voice. *Arias and songs by Gounod, Lehár, Orjf, Puccini, Rachmaninov, R. Strauss, et al. Renée Fleming, soprano; English Chamber Orchestra, Jeffrey Tate, conductor. 289 '458 858-2 (1997).*

Critics seldom point out the difference between a beautiful voice and a *very* beautiful voice; the music lover has to do some independent listening. The title of this CD should include that "very," since Renée Fleming's voice is of the very highest order—Eleanor Steber comes to mind as a soprano who had a similar effect on me in my younger years. Her musicianship is also impeccable; her sense of style is highly sophisticated; the only thing missing here is music that I truly love. (I can live without the "Jewel Song" from *Faust* and stuff like that, but then she also has Mozart and Schubert CDs on the same label.) As for the recorded sound, it could be a wee bit more flattering to her top notes but is quite excellent on the whole.

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Richard Strauss: *Also sprach Zarathustra, Op. 30; Till Eulenspiegels lustige Streiche, Op. 28; Salome: Tanz der sieben Schleier, Op. 54. Berliner Philharmoniker, Sir Georg Solti, conductor. 452 603-2 (1996).*

My obsession as an *Also sprach Zarathustra* "compleatist" is richly paid off here—this is the finest recorded performance, in my opinion, since the 1954 Reiner/Chicago classic. Reiner's phrasing of the string passages is still the most powerful and riveting I have ever heard, and the groundbreaking RCA Victor Living Stereo recording is obsolescence-proof, but the Solti/Berlin combo is almost equally persuasive. Splendor and lyricism are in perfect balance in Solti's interpretation, and the subtle inner details of the orchestration are illuminated as never before. The virtuosity of the Berlin players is stunning and further displayed in *Till* and Salome's dance, whose renditions are in the same class with *Zarathustra*. The sound, recorded in live concerts in the orchestra's home venue, is actually better than standard English Decca (see above), with superior soundstaging, terrific bass, and a sweet top end. James Lock, a name new to me, was the engineer. A great CD—check it out.

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Richard Wagner: *Die Meistersinger von Nürnberg. José van Dam, Hans Sachs; Ben Heppner, Walther von Stolzing; Alan Opie, Beckmesser; Karita Manila, Eva; Herbert Lippert, David; Iris Vermillion, Magdalene; René Pape, Pogner; Chicago Symphony Orchestra & Chorus, Sir Georg Solti, conductor. 452 606-2 (4 CDs, 1995).*

If performance and audio quality are equally important to you, and you own only one recording of *Meistersinger*, this should be the one. Probably the last major recording by Solti while he was still in full possession of his powers, it satisfies all the top-priority requirements: good singers, good orchestra, good conductor, good concept of what Wagner's music is all about. (That any composer could create this stupendous celebration of the diatonic scale right after staging the chromatic revolution with *Tristan* is one of the miracles of musical history.) Solti applies a light touch to the many monologues and dialogues in the opera, never letting the orchestra drown them out (he explains this in an introductory program note), but he hasn't forgotten how to unleash the almighty power of the Chicago Symphony in the big moments. Heppner and van Dam are about as good a Walther and Sachs as you can find these days, and the rest of the cast is also splendid. The Decca team from England did a very creditable job compiling the finished product from several live performances in Orchestra Hall (which today is far from the perfect recording venue it was back in the days of Lew Layton); the sound is more than good enough not to be an issue when opting for this particular version of *Meistersinger*.

Mapleshade

See my previous comments (Issues No. 22 and 24) on Pierre Sprey's mind-boggling results with his live-to-2-track analog recording technique.

"Brand New Bag." *Ebony Brass Quintet with Hamiet Bluiett. MS 03032 (1994).*

Most lifelike recorded brass sound on the planet. A gigantic tuba, a French horn, a trombone, and two trumpets play avant-garde jazz (almost Webern-like) right in your face. A totally awesome experience.

*"Makin' **Whoopee"** (Tribute to the King Cole Trio). King/Bluiett Trio: Hamiet Bluiett, baritone sax; Rodney Jones/Ed Cherry, guitar; Keter Betts, bass. MS 04832 (1996).*

Jazz as I like it, 1950s smoke-filled barroom style. Bluiett has a highly individual style on the stentorian baritone sax, and Pierre Sprey's recording is you-are-thereness incarnate.

MusicMasters Classics

This label represents the most uncompromisingly serious side of the BMG Music empire. It sure works for me.

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***Igor Stravinsky:** The Composer, Volume IX. The Firebird (1910) plus 5 shorter works. The Philharmonia and London Phiharmonic Orchestra (the latter only in 2 short pieces), Robert Craft, conductor. 01612-67177-2 (1997).*

Epiphany! All the little corruptions and mistaken edits that have crept into Stravinsky's original 1910 score over the years have now been cleaned up by a team of German scholars, and the world's greatest Stravinsky authority (indeed, Stravinsky's alter ego) is performing the authoritatively restored, pristine score here with a great orchestra. What more do you want—Pierre Boulez? I find Craft sufficiently inspired in the Stravinsky repertory to keep me from craving a virtuoso superstar conductor in this wonderful music. Craft conducts the corrected score correctly, the way Stravinsky would have, and that's all it takes for a great *Firebird.* The recording, produced by Gregory K. Squires like the previous eight volumes, is perhaps the best-sounding of the series, possibly on account of the London hall. Close-miked detail, brightly etched but never to the point of unpleasantness, is combined with panoramic soundstaging and a huge dynamic range. The bass line, so important in this score, is particularly impressive. This is clearly the "canonical" *Firebird* for the nonce. (Oh yes, the fillers are also great stuff.)

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W. A. Mozart: String Quartets, Vol. TV(K. 172, K. 168, K. 458, K. 590). The American String Quartet: Peter Winograd & Laurie Carney, violins; Daniel Avshalomov, viola; David Geber, cello. 01612-67171-2 (1996).

See my comments in Issue No. 24 on the first three volumes of this outstanding series. The matched set of Antonio Stradivari instruments from the Smithsonian Institute sound as wonderful here as ever, and the quasi-Beethovenian Adagio of K. 458 (the "Hunt" quartet) alone is worth the price of admission. Not that K. 590 is anything to be sneezed at. What's more, Judy Sherman's recording technique with this group seems to be getting better and better.

Nonesuch

Any label that records Richard Goode is good. (A deliberate Gertrude Steinism by your Editor.)

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***Frédéric Chopin:** Polonaise-Fantaisie in A-flat Major, Op. 61; Nocturne in E-flat Major, Op. 55, No. 2; five Mazurkas; Scherzo in E Major, Op. 54; Barcarolle in F-sharp Major, Op. 60. Richard Goode, piano. 79452-2 (1996-97).*

At this point in his artistic development, Richard Goode is one of the world's great pianists. His greatness comes through no matter what he plays. He is definitely not a Chopin specialist, but in this case that's actually a good thing. He addresses the often trivialized composer with a Beethovenian seriousness, playing the notes as written, taking very few liberties, using less rubato and pedal than the flamboyant Chopinists, but achieving tremendous clarity and a thoroughly convincing poetic quality all his own. It isn't traditional Chopin playing but it's great. I am particularly impressed by his inclusion of a large assortment of Mazurkas in his first recorded Chopin program; even Rubinstein found those quirky rhythms challenging. Easy success is not what this artist seeks. The piano sound as recorded by Max Wilcox is superb. My only quibble is the mere 48 minutes of music on the CD—if Richard Goode ran out of Chopin pieces he felt comfortable with, he could have given us, say, a little Schubert.

RCA Victor Red Seal

The classic of classics among record labels (now owned by BMG Music) has finally achieved sonic parity with the audiophile brands, while continuing to employ better artists.

***Ludwig van Beethoven:** Fidelio. Deborah Voigt, Leonore; Ben Heppner, Florestan; Elizabeth Norberg-Schulz, Marzelline; Michael Schade, Jaquino; Matthias Hblle, Rocco; Günter von Kanneri, Don Pizarro; Thomas Quasthoff, Don Fernando. Bavarian Radio Symphony Orchestra and Chorus, Sir Colin Davis, conductor. 09026-68344-2 (2 CDs, 1995).*

Beethoven was probably the greatest composer who ever lived, but this is not a good opera, no matter how politely musicologists and critics speak of it. The vocal writing is heavy-handed, formulaic, lacking in grace, and unflattering to both male and female voices. The orchestral writing rises to Beethovenian heights here and there (how could it fail to?), but not to the level of the contemporaneous symphonies. Beethoven kept revising the work; the final version is arguably the best but still clunky, saved only by the tremendous sincerity of the great composer's effort and the drama of the famous offstage trumpet fanfare that everybody waits for. If *Fidelio* were an opera by the other Ludwig of the era, Spohr, I wonder how often it would be performed. Once every 50 years? This recording is about as good as can be put together anywhere today—one great singer (Heppner), several very good ones (especially Voigt), excellent orchestra and chorus, and a world-class conductor who knows how to bring out the music's strengths and lighten up on the weaknesses. The Leonore Overture No. 2 is included (maybe because it isn't as much of a warhorse as No. 3). The sound quality is beyond reproach.

***Sergei Prokofiev:** Romeo & Juliet (Scenes from the Ballet). San Francisco Symphony, Michael Tilson Thomas, conductor. 09026-68288-2 (1995).*

Prokofiev's best music (in my opinion), magnificently played and so well recorded that this is now one of my demo CDs. The suite is not one of Prokofiev's but a new one, arranged by the conductor, but all the familiar music from the ballet is there. I couldn't offhand name a more convincing and natural-sounding recording of the modern symphony orchestra. The producer was Andreas Neubronner, the engineer Markus Heiland, in the first collaboration between RCA Victor and Thomas/San Francisco. Get this CD if you don't already have it.

Telarc

This label, famous for its sound, is beginning to run out of standard classical works to record and appears to be leaning more toward jazz and other upscale popular music.

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***Ludwig van Beethoven:** Fidelio. Gabriela Benacková, Leonore; Anthony Rolfe Johnson, Florestan; Siegfried Vogel, Rocco; Franz-Josef Kapellmann, Don Pizarro; lldiko Raimondi, Marzelline; John Mark Ainsley, Jaquino; David Wilson-Johnson, Don Fernando. Scottish Chamber Orchestra and Edinburgh Festival Chorus, Sir Charles Mackerras, conductor. CD-80439 (2 CDs, 1996).*

Please read my comments on the music itself in the review of the RCA Victor recording above. This version is more interesting sonically because of Mackerras's use of natural horns and small-bore brasses—a more authentic early-19th-century sound, very vividly recorded by Jack Renner, in the best Telarc tradition. The conducting itself is crisper, more incisive than Davis's, but maybe too much so—Mackerras has something of an agenda. The singing is not in the same league with the RCA version; it is quite undistinguished across the board and ultimately the decisive factor. The Leonore Overture No. 3 is included as an appendix.

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***Gustav Hoist:** The Planets. Atlanta Symphony Orchestra, Yoel Levi, conductor. CD-80466 (1997).*

Most audiophiles have several versions of this showy warhorse in their CD collection. Should this one be added? The Michael Bishop 20-bit recording is perhaps the widest in dynamic range and the most clearly delineated texturally of any version, but I think Levi's conducting is too rigid and ungenial for the essential character of this music. My favorite recording of the digital era is James Judd's with the Royal Philharmonic on Denon (1991). It is so much more relaxed and lyrical than Levi's and even the sound is more sumptuous, though a bit less close-up, in a much better hall. The Telarc CD is labeled Surround Sound with the explanation that it is compatible with all surround sound systems. Huh?

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***Gustav Mahler:** Symphony No. 6 in A Minor ("Tragic"). Atlanta Symphony Orchestra, Yoel Levi, conductor. CD-80444 (1997).*

The English critic Burnett James thinks this is Mahler's best symphony. The American composer and critic Paul Turok thinks this is the best-ever recording of the symphony except for the old Karajan/Berlin analog set. Who am I, ex-adman and plodding audio journalist, to contradict these experts just because I've been a music lover all my life and have a good stereo system? I can think of three or four Mahler symphonies I like better, and (just as an example) the nine-year-old Chailly/Concertgebouw recording on London is played with greater orchestral refinement, but Levi does combine elemental power with great clarity. The Jack Renner 20-bit recording, labeled Surround Sound with no explanation, has tremendous dynamic range (dig those famous hammer blows!) but tends to be a bit too hot on the top end in the louder passages.

Teldec

The following is not a CD but a videotape, which I hope will be available on DVD one of these days. It has also been issued on LD, but that's a moribund medium at this point.

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***"The Art of Conducting—Legendary Conductors of a Golden Era."** Sergiu Celibidache, Wilhelm Furtwängler, Erich Kleiber, Willem Mengelberg, Evgeny Mravinsky, Charles Munch, et al. 95710-3 (released 1997).*

This is a sequel to *"The Art of Conducting—Great Conductors of the Past,"* which was released in 1994 and which I have seen but not reviewed. The sequel presents longer passages featuring fewer conductors; both are invaluable historical documents. Do not expect high fidelity, just the most intimate contact with unforgettable musical presonalities. •

Book & Software Reviews

By the Staff of *The Audio Critic*

High Performance Audio Power Amplifiers
By Ben Duncan
Newnes (an imprint of Butterworth-Heinemann) 1996, 479 pages

Audio Power Amplifier Design Handbook
By Douglas Self
Newnes (an imprint of Butterworth-Heinemann) 1996, 329 pages

Reviewed by David Rich

Here are two books on the same subject, published by the same U.K. publisher with the same publication date, yet they are as different as black and white (hats?). You will get the point right from the start when you read each author's introduction. On page 1 of the Duncan text we are presented with a "Hip Bootable" quote from John Atkinson, who is introduced as the "foremost international writer on, and reviewer of, audio quality." This is followed by definitions of airy, closed in, pace, rhythm, etc. (5 pages of this stuff). Then we move on to a section about psychoacoustics which includes the statement that frequencies up to 80 kHz can be perceived by the brain.

Switching to Chapter 1 of the Douglas Self text, we find on page 5 a section titled Misinformation in Audio: "In the last twenty years the rise of controversial and non-rational audio hypotheses, gathered under the title *Subjectivism...*"" There follow 16 brilliant pages on the realities of audio design and the failures of the subjective analysis. Douglas Self concludes Chapter 1 with a section on the performance requirements for an amplifier—safety, reliability, power output, frequency response, noise, distortion, etc. It will all be very familiar to the reader of ***The Audio Critic.*** What follows are 13 chapters on how to design power amplifiers that achieve high performance against real, not imagined, performance specifications.

The Self text may be a bit of a rough go for non-E.E. types. It is written at a level that assumes familiarity with analog electronics at the junior level of an E.E. program. This is a higher level than what I try to write at in this publication, but if you understand my stuff you should get a lot out of the book. I do wish that the text were a little more tutorial, with more complete explanations of the basics and more complete derivation of equations. The added material would have allowed the book to be understood by an audience that desperately needs to be shown the truth about amplifier design. The text is well documented with annotated schematics, computer simulations, and lab measurements. This is a book of science and it makes clear why a grainy midrange, etc., is science fiction.

If you do slog your way through the text you will be well rewarded. Eight different distortion mechanisms of a standard-topology amplifier are identified and explained. Some distortion mechanisms are surprising by their simplicity, such as taking the feedback connection off the wrong point at the amplifier output. Audiophiles may be surprised to see a discussion of power-rail coupling distortion and distortion from capacitors. Some parts of audiophile lore do have grounds in science, but these effects are measurable and easily correctable. Audiophiles may find it satisfying to learn that class A amplifiers are indeed more linear, but chagrined to find that increasing the bias of a class B amplifier makes things worse.

Much of what has been published in ***The Audio Critic*** is developed independently in Self's text. BJTs are preferred over JFETs at the input stage and over power MOSFETs at the output stage. Current mirror loads are prefered over resitors in the first stage. The advantages of placing a follower between the first and second gain stages are explained clearly. The major disadvantages of linear regulated power supplies are nicely dealt with (take that, Dan Banquer). Instead of adding Band-Aids to the power supply, Self discusses, in detail, methods to make the amplifier more robust with regard to power rail fluctuations. About the only thing on which I really disagree with him is that he calls slew rate an amplifier's "speed."

The only real downside of the text is its concentration on one circuit topology. The impression is given that this topology is the optimal one, when in fact it is one many excellent topologies. While one can understand the author's desire to limit the size of the text by limiting discussions of alternative topologies, it would have been nice to see coverage of high-performance designs in more depth. Important omissions are a significant discussion of topologies with three voltage-gain stages that work well in many Japanese amplifiers, discussion of alternative methods of output stage protection besides *VI* foldback limiting, and a discussion of output stages that use dynamic bias stages. It is also surprising that the advantages of degenerating the second gain stage are not presented by Self, even though they were clearly demonstrated by Edward Cherry in the early '80s. Also, Self's dismissal of compensating an amplifier by adding a capacitor to ground on the second gain-stage output, while very plausible, fails to explain why so many excellent designs use this approach.

You get lots of different circuit options discussed in Duncan's book, which is more of a survey text. Unfortunately, Ben Duncan is a card-carrying member of the high-end audio party, and much of what is in this book would also be right at home in *The Absolute Sound.* He is on better behavior here than in his *Hi-Fi News & Record Review* column, where he has talked about how solder affects sound quality, but he is far less rigorous than he has been in his articles for *Studio Sound.* His analysis of the different topologies in this text often enters the land of voodoo. The sonic quality of different stages is discussed without any justification of why they should have a sound at all. Often we get a little English jingoism thrown in, as in this example on dc servos: "Servos have been *de rigeur* in US and US influenced. ..amplifiers...Tellingly, servos are not usually nor likely to be found in amplifiers with truly accurate sounding bass." As expected, the Japanese are not left out of the author's biased view of the world: "...the history of class G demonstrates how the Japanese giants have tinkered with audio technology, while the smaller US companies 'got stuck in'."

The text in some ways can be recommended for the unintentional comic effects. For incoming QC inspection of bipolar output devices we are told to test for: V_{BE} for matching, beta, noise, breakdown voltage, and sonics. We are told that "individual BJTs maybe listened to, typically in a simple circuit (e.g. class A, 1watt into 8Ω) fed by a high resolution music source." It comes as no surprise that Duncan favors MOSFET output devices despite measurable problems, to wit "there is usually some un-nullable crossover residue yet sonics are perceived as benign or absent..."

Duncan does give some technical analysis in his section on MOSFETs to go with the mysticism; for example, he cites the wider bandwidth of power MOSFETs and reports that the high R_{0N} of a MOSFET is "a relatively minor matter" as a disadvantage. But trying to bend the facts to fit an agenda is fraught with peril, and all we have to do is go to Douglas Self's book to find out what's wrong here. Self points out in his book that the high R_{0N} is going to "make more likely the lowering of the output pole by capacitative loading," thereby preventing the dominant pole of the amplifier from being raised, even given the MOSFETs' wider bandwidth. Self also points out that power FETs are difficult to parallel because of wide parametric variations and a tendency to go into parasitic oscillation, in direct contrast to Duncan's incorrect claim that they are easy to parallel. He then goes on to present an analysis and computer simulations showing that BJTs are more linear than FETs in class A and class B operation when transconductances are matched. In a court of law, Self's hard science would easily beat Duncan's marginal technical analysis and single-presentation listening-test results.

It comes as no surprise to find that Duncan comes down on the side of regulating power supplies: "Another approach is to use electronic (i.e. active) means to reduce the supply impedance. Sonically, this the real benefit of *regulated* supplies." I recommend that Mr. Duncan get himself a copy of Douglas Self's book so he can see the better and cheaper methods discussed in Chapter 8 of Self's text for improving power supply rejection. We again should not be surprised to find Duncan's position on feedback: "...for music signals, where there is little continuity, NFB is condemned to be *always* trying to catch up with what has just happened." Self neatly disposes of this and a number of other feedback myths—including the one *Stereophile* is currently promoting, that NFB increases higher-order harmonic energy—in his section titled Some Common Misconceptions about Negative Feedback.

I should point out that the Duncan text does have a significant amount of useful information, but you must be a sufficiently well-informed reader to be able to separate the facts from the audiophile fantasies. Mr. Duncan is an audio amplifier designer and not one of those history professors (or whatever) who do audio reviewing as a hobby. It is fascinating to watch him deal with the design issues when the realities of engineering clash with the audiophile belief system. An excellent example is when he explains the need for a series inductor at the amplifier's output. "The motto *'no component unless essential'* is admirable, but inductorless power amplifiers are renowned for becoming RF unstable..." If you must have every book in print on amplifier design, go ahead and add Ben Duncan's text to your collection, but most readers of ***The Audio Critic*** are clearly much better served by Douglas Self's wonderful seminal text, which I highly recommend to anybody willing to spend the time to understand it.

The Home Theater Companion
Buying, Installing, and Using Today's Audio-Visual Equipment
By Howard Ferstler
Schirmer Books (an imprint of Simon & Schuster Macmillan) 1997, 450 pages, $40.00

Reviewed by Peter Aczel

The title and subtitle of this big fat paperback are strangely misleading. They actually underplay the scope of a tremendously ambitious one-volume effort. "A Comprehensive Introduction to Audio and Home Theater for the Diligent Novice with a Long Attention Span" would be a more appropriate label for the author's intentions and the book's contents. Literally everything is covered, from the meaning of a watt to the dismissal of faddish tweaks, with long explanations of all the current technologies, including the very latest. Howard Ferstler must have toiled like no other writer on consumer electronics known to me.

Considering the breadth of information in the book, the Johnsonian "it is not done well, but you are surprised to find it done at all" comes to mind. Actually, a great deal of it is done well—well enough to make it unquestionably recommendable to the intended readership—but there are a few scientific bloopers and tutorial misfires which unfortunately, and perhaps unfairly, hurt the credibility of the total text. Ferstler is no E.E. or physicist but he comes on like one.

For example, Ferstler states that Sony Super Bit Mapping can extract "true 19- or 20-bit performance" from a 16-bit PCM-encoded CD, thus confusing psychoacoustic tricks with quantization noise. He also thinks that so-called 18- and 20-bit D/A converters make the external trim pot adjustments less critical, thus ignoring (a) that currently manufactured DAC chips cannot be trimmed externally and (b) that those 18 and 20 bits are in nearly all cases "marketing bits" with little relevance to actual resolution. He furthermore thinks that Roy Allison is God, a ranking not quite borne out by the performance of RA's speakers. And that's just a random sampling of questionable items.

Leafing through the book I had

occasion to be amused by Ferstler's essentially 1960s-oriented mindset, which runs in counterpoint to all the up-to-date information. Some of his illustrative examples surely belong in the Smithsonian. Was the Edgar Villchur era truly the belle époque, Howard?

Don't misunderstand me, though; it's just that Howard tends to bring out the nitpicker in me. Blemishes aside, the book is fundamentally sound, as easy to read as a magazine (wide margins, icons, sidebars, charts, etc.), and basically on target for the slightly nerdy tyro.

The Complete Guide to High-End Audio
By Robert Harley
Second Edition
Self-published 1998,558 pages, $29.95

Followup by Peter Aczel

The original 1994 edition of this grotesque exercise in untutored techie-exegesis-cum-subjectivism was reviewed at some length by David Rich in Issue No. 24. He discussed about a dozen unprofessional technical blunders in the book and then gave up on the rest of it. He is not interested in going over the same boring and depressing ground all over again just to review this emended and expanded new edition, nor am I except for some very general philosophical observations.

David does have, however, a mea culpa to insert here (which he also posted on the Internet).

> I wrote that Harley "then moves on to explain that a differential amplifier converts balanced signals to unbalanced signals—wrong again. Trust me, this is what he thinks, as Figure 5-9 shows an XLR input connected directly to a block he marks Differential Amplifier." Well, any college student who has taken his first electronics course would tell you that Harley's statements are absolutely correct and I am totally wrong to object to the them. I apologize to Mr. Harley for this major blunder. I hold that my other statements in this book review are correct but this embarrassingly basic error reduces the impact of the ideas I was trying to convey. In my rush to explain the many errors in the book I pulled in correct statements as well as the errors. This sloppy reporting is inexcusable.—*David Rich*

Now, in my humble opinion,if Robert Harley were a mensch like David Rich instead of an insecure and spineless lightweight, he would also have openly and contritely apologized for his not one but myriad dumb-ass errors, perhaps even diffidently offering David a freelance assignment as the fact-checker for the second edition (which David would then have politely declined). Instead, here is what Harley did:

The circuit schematics with screwed-up explanations that David specifically pointed out in the first edition are gone in the second edition. Just gone—no schematic, no explanation. Some totally erroneous statements found by David that were possible to correct by changing a word or two are now correct. Even so, most of the original errors, and especially those not discussed by David, have been retained. In some cases an error corrected in one chapter resurfaces later in the book. It's fairly obvious that David was Harley's unacknowledged and unrewarded fact-checker, but of course the mooched corrigenda did not extend from cover to cover, and that remains the big problem. David should have finished the job—right, Bob?

Thus *The Complete Guide to High-End Audio* in its second edition is still disastrously flawed as a sourcebook. Perhaps its only value is as a sociocultural document of an era in which incompetent instruction finds a ready market, especially when the instructor is some kind of cultist.

ETF4.0 Room Acoustic Measurement Software
ETF, Oshawa, Ont., Canada, $199.95

Reviewed by David Rich

If you have spent four figures on your hi-fi system and you have room to move your speakers around and/or put room-treatment material in your listening room (such as the Echo Busters discussed in the last issue), then you need this software. It is best to state what it is not—it is not software for designing and measuring loudspeakers. What it does is measure your loudspeaker *and* room. And when you see what you are really listening to, you will know that wire is not going to make any difference in the sound of your system.

What you need, of course, is a computer, and it has to be in your listening room. If it is a laptop, no problem. If it is a desktop and in the same room, you are still OK, but it cannot be in another room. What you do not need is any special hardware. ETF uses the ADCs and DACs built into your computer. Douglas Plumb, who wrote the software, is I believe the first and only person to come to this realization and make it a practical solution in software for acoustical measurements. Everybody else requires add-on electronics (so much for laptops), and they charge four-figure prices. One must note that using internal data conversion does come with the price of working in a noisy environment. This can affect measurement accuracy but not enough to be significant for the purpose here.

Any computer made in the last three years should have a good enough sound card. Call ETF if you have doubts. For an extra $200, ETF will sell you a calibrated mike and preamp, but Radio Shack stuff is good enough to get you started in seeing relative differences. The computer's ADC encodes the mike or line input (depending on the gain you need). The computer's headphone (or line) outputs are connected to your stereo aux inputs. The DAC generates the test tones. Is this clever or what? Obviously you need to take care with anything that generates large-scale test tones. Keep the volume low until you know what level the signals are at.

I could take pages to show you all the things you can do with this, but given the fact that a free preview version is available on the Web **(www.etfacoustic.com)** and that a couple of large reviews have already appeared (again see the Web site), I shall be brief. I hear some of you out there saying you have no access to the Web. OK, back to audio reviews. The free Web version is actually very useful, with a surprising amount of the program features left intact. To use ETF you must have a computer, and in all likelihood you are on the Web. (Another incentive to be brief is that your Editor thinks major rearrangements of a room are allowable only in bachelor pads or dedicated listening rooms, and thus he is not giving me much space.) Among the things the software generates are low-frequency response and distortion curves. It also generates a variety of energy/time curves taken over different frequency bands. Frequency response plots with different time gating and 3-D frequency response curves (waterfall plots) for identifying resonances are also generated. Unless you know your computer is free from digital noise (definitely not the case when it comes to my laptop), I would not trust any THD reading below 1% (a nonissue in this application).

Data crunching is a little slow on my low-end Compaq 1210 laptop, but waiting for results is part of the fun, at least when you first use the thing. After that, the two-minute wait gets to be a drag. Ground loops are easy to form with computer peripherals. If you hear hum, yank the printer cable out of the computer. Even this PC-illiterate analog-loving creature found the user interface to be intuitive and easy to use. The only point of frustration I had, besides finding the darn ground loops, was in setting up my sound card's control panel.

The nifty online instruction manual gets you started in the subject of room acoustics and how to improve things. If you really get into this, it is off to the (online) book store to pick up a text on room acoustics and treatment. Understand that you are an amateur in a world that requires a professional to get the job done quickly and correctly. The key thing to take away from this review is that this software is for room acoustics measurements. There is no mode that subtracts the room out of the measurements and leaves you with a loudspeaker measurement. Think of it as computer-assisted tweaking that can make a real difference. The more you read and spend time using the software, the better you get at it. It's lots of fun if you are into nerdy things like this—but not if you're not.

RPG Room Optimizer™
RPG Diffuser Systems, Inc.
Upper Marlboro, MD
$495.00

Reviewed by David Rich

This software just arrived at the last minute. Kevin Voecks (Revel Speakers) recommended it to me when I was discussing the problem of finding the optimum speaker placement with the ETF software. The ETF software measures your room and helps you determine optimal placements by trial and error. The RPG software tells you where to start, so you can spend less time doing the final optimization. RPG is a company that makes room treatment not only for homes but also large acoustic spaces and studios. The Web site (**www.rpginc.com**) shows an impressive list of clients. This is not a tweako high-end company but a professional organization providing products for acoustical engineers and installers. You can also download a demo program from their Web site. The demo lets you play with the program for a preset room size. Unlike the ETF demo program, the RPG demo gets you interested but provides no functionality to allow you to optimize the room you live in. To optimize your room you have to purchase the program. Warning—once you play with it, you will want the real one for yourself and it is not a cheap program, so download with this caution in mind.

The program calculates under constraints the recommended placement of your loudspeakers and of you the listener, using computer optimization. The constraints are set in a very easy-to-use series of setup panels (advanced features require a deeper commitment to learn the program and the fundamentals of room acoustics). You tell the program how big your room is (rectangles only at the moment), which wall you want to place the speakers on, what limits you impose on where the speakers will be placed in the room, and what limits you impose on where you will be (including ear height). The program needs to know where the woofer is with respect to the floor. If you want to, you can let the program put the speakers on virtual stands and it will determine the height of the stands. The program knows how to deal with two woofers in a speaker. It knows how to work with subwoofers and it will do stereo or 5.1 (5 matched speakers or THX dipole) configurations. While trial and error may get you close to optimum in a two-speaker deployment, with 5.1 it is impossible. It is important that the program has the ability to constrain the speakers' position because the best solution for bass response will not always be optimal for stereo imaging and other factors (like where your significant other says the speakers can go). The program also tells you where to put acoustical room treatment by calculating the points on all the walls (yes, floors and ceiling included) where the first bounce position occurs for all loudspeakers..

For computer science types I will mention that the simplex routine is used to carry out the optimization process. The simplex routine is a robust optimizer for linear systems. Other algorithms may be faster but not as robust. The cost function of the optimization is the evenness of the response the listener hears. The program allows you to watch the optimization process in action. You see virtual speakers and listeners fly around the room as the program looks for the best position. You can also watch the spectra change as the program attempts to smooth out the response. It takes a while to do this, running up to 500 trials in the default setting. The program does try to keep stereo imaging in mind by keeping the ratio of the distance from the listener to the center point of the speaker plane and the distance between the stereo pair in the range of 0.88 to 1.33. This nonlinear constraint creates complexity for the chosen optimization method. Dr. Peter D'Antonio and Dr. Trevor J. Cox, who wrote the program, report in a journal-like paper supplied with the program (revised from AES Preprint 4555) that geometric constraints increase the chances of finding the best (global) minimum for the problem under optimization with the nonlinear stereo constraint. Information on this and much other detailed information can be found in the AES preprint.

I believe this is the only program that takes both the speaker-boundary interface effects and modal excitation response into account, and the program's algorithm attempts to optimize both. Modal response results from the natural resonances that occur as sound waves reflect back and forth between solid walls. The effect is minimized by placing the speaker in the room in such a way that it is minimally affected by the room's natural modes. Speaker-boundary interface effects are different. They occur through the interaction of the direct sound from the loudspeaker and reflections from the walls. The so-called Allison effect is one example. The program works from 300 Hz on down, where both these effects are dominant.

Answers are generated by the program in feet and inches or meters. The results are two decimal places deep. One wonders why such a high accuracy is specified. No sensitivity functions are supplied at this point in the program's development. Thus a small change from the recommended placement may make a large change in room response. The program does not consider sensitivity in its optimization. The optimization itself was robust to all examples I tried (remember, this is a last-minute review).

At $495 the software is pricey. Special-purpose software costs a lot to develop and support. The company says that it is working on optimization for multiple listening positions, nonrectangular rooms, and even software that lets you optimize your room size (the ultimate tweak, I would think). The Room Optimizer is clearly aimed at dealers and installers given its price, but that should be not be a deterrent if you have five figures invested in your system. If you use the program and follow its advice, you *will* get better response from 300 Hz downward. Cable will not change your system's sound; new electronics will not change it for the better (single-ended triodes will make it worse); but this software will improve the sound you hear. I should point out that any dealer/installer should be using this software and a room analysis program such as ETF. That is what those 50 points are for. If they are not using these tools, but are instead doing installations with the help of only their golden ears, they are committing acoustical malpractice. •

How to Be a Sophisticated Audiophile *(continued from page 40)*

statement about them; there may be small differences in video quality; the audio performance appears to be perceptually equal to that of ordinary CD players.)

As for outboard D/A converters, they have no inherent advantage over built-in ones except that they generally have multiple digital inputs (coax, optical, and sometimes AES/EBU, ST, etc.) for multiple digital sources. That is totally irrelevant to consumers with one CD player and no other digital source, or even to those with AV electronics equipped with multiple digital inputs. With an outboard D/A converter, jitter at the digital interface becomes an issue, although it is hardly ever a significant problem from the user's point of view; a built-in DAC by its very nature steers clear of that risk. I am not implying that you should avoid an outboard unit; just have a damn good reason for getting one.

What you should avoid like the plague are tweako/weirdo accessories and add-ons, such as ebony wood pucks or magic bricks that you put on the equipment to make it sound better, or cable burn-in devices, or anything to treat your CDs with, or horrendously costly power conditioners, or clocks you plug into your wall for better sound, or pure silver interconnects—the list is endless, but the common denominator is the lack of a scientific raison d'etre. Sometimes we publish specific "gotcha" commentaries on these frauds, but there are just too many of them. The rule of thumb is that if it looks like snake oil, smells like snake oil, and is promoted like snake oil, then it *is* snake oil. Put your hard-earned money into better speakers, for heaven's sake.

All of the above is more common sense than rocket science. The trouble is that pure common-sense information has become extremely scarce in the audio world. Even those who are completely aware of the difference between fact and fiction tend to equivocate under the pressure of marketing forces. For example, why hasn't every audio journalist who is not an idiot denounced the wire and cable ads as fraudulent? Because the checks collected by the journalists from their publishers are drawn on the money the publishers make from those ads. You don't bite the hand that feeds you, even if it isn't clean. Thus the simple truth is revealed only in private conversations, if at all, and very rarely in print. (Note that I said "every audio journalist who is not an idiot." There are a few who actually believe all the tweako articles of faith. Note also that this publication carries no advertising for products that defraud the consumer, but then we are not even tempted on account of our low overhead.)

One Last Mystery

What puzzles me is why there is so much more B.S. in audio than in, for example, the automotive or photographic fields? Why aren't we told to use silver wire in our car's ignition system or to paint the outside of each roll of 35-millimeter film with a special green paint? Maybe some of our readers can answer that. •