The Audio Critic®

Issue No. 22
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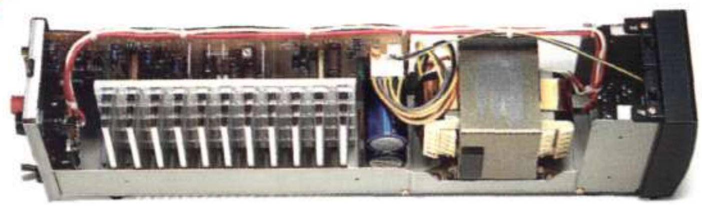

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In this issue:

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In an editorial that will raise some eyebrows, and even some blood pressures, your Editor analyzes the hypocrisy of the high priests of the High End.

We continue our survey of amplifiers and preamps.

The question of when, if ever, absolute polarity is audible is clarified in a letter from a top authority.

Plus other test reports, all our regular columns, and the return of our popular CD capsule reviews (oodles of them).

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From the Editor/Publisher:

This issue is dated Winter 1994-95. The last issue, No. 21, was dated Spring 1994. That was late spring; this issue goes to press in early winter, so the gap is smaller than the apparent nine months but bad enough. What are we doing about it? Lots of things—but we have learned, painfully, not to make promises before the implementation is a reality. Three things are certain: (1) something has to give; (2) we are here to stay, regardless; (3) we are not changing our editorial stance. That still leaves a number of viable scenarios to choose from.

Box 978

Letters to the Editor

An archetypal letter we seem to get again and again lists in loving detail all the components in the writer's system, down to interconnects and tiptoes. In nearly every case it's quite unclear what the letter writer wants. Our official blessing? Recommended changes? Recognition as a blood brother? Please, all you audio addicts, if you insist on talking about your equipment and want to get our attention, make sure you explain how it all ties in with our editorial concerns. Letters printed here may or may not be excerpted at the discretion of the Editor. Ellipsis (...) indicates omission. Address all editorial correspondence to the Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951.

The Audio Critic:

...I have read Issue No. 20 and found it quite charming. Your elegant chopping to pieces of the audio tweakies was very nice indeed. I even enjoyed the writings of the others, many of whom I know quite well....

* * *

[Six weeks later:]

I have now read the several issues of *The Audio Critic* which I recently received. They are very interesting, and I want to congratulate you on getting some very good people to write for you. I am a bit surprised that you and some letter writers refer to my modest work so frequently but appreciate the interest I have generated. There are many issues that need deep thought, and I am delighted to have these discussions take place, since the light envoked (with some heat) eventually seems to pry out the truth.

It is very annoying to have persons at the extreme fringes of an issue use one's writings to prove their point with elements of these writings taken out of context. Unfortunately, it happens only too often. Shades of gray are too often made black or white by fanatics.

I was pleased to see complete paragraphs from my paper quoted in context printed in Issue No. 21. Additionally, your interpretation, printed on page 8, of what my article said is quite accurate but a bit more truncated than I would have preferred. If you have the patience, I supply for you herewith my own summary of my work on acoustic polarity.

For perspective, my paper on polarity was presented at an AES convention in 1991 and was published in the *Journal of the Audio Engineering Society,* Vol. 42, No. 4, 1994 April. Needless to say, this is a professional Journal for which all articles are extensively reviewed for quality and accuracy. A version of this paper was also printed in *Audio* magazine with minor modifications to satisfy a different (and much larger) audience. Nevertheless, the essential contents of the several versions of the paper are identical.

I strongly suggest that anyone interested in this matter read the original paper and not someone else's interpretation of what it might say. Some of the listening tests are fairy easy to duplicate, and the paper is written in very simple, not highly technical, terms.

These comments cover both the work I did in 1991 and my more recent experiences (1993-1994), which are indicated by brackets [...].

There are four points made:

1. It is clearly possible to show the audibility of acoustic polarity inversion with steady-state tones (electronically generated) or quasi-steady-state tones (produced on physical instruments but with steady monotone playing techniques). These are boring, nonmusical tones. The audibility of acoustic polarity inversion for these very special cases has been documented by several authors. [I have repeated this listening experience many times with both headphones and various loudspeakers, and the experience is so definitive that there is no question about its existence.]

2. When real musical performance material is used in such tests, it is very, very difficult (nearly impossible) to hear the effects of polarity inversion. Our large group tests showed only very slight positive results with loudspeakers in a highly idealized and simplified listening environment. [These listening tests have been repeated with headphones and a great variety of loudspeakers, and it has been confirmed that it is very, very difficult to hear polarity inversion. Nei-

ther I nor anyone I know, and trust, has heard acoustic polarity inversion with stereo program material in a normal listening environment.]

3. Because it was so easy to hear polarity inversion with simple steady-state tones and so difficult to hear with real music, a large part of the paper is devoted to trying to determine the reasons why this is the case. A major part of the paper suggests, but does not firmly define, these reasons. More work is required to define this very subtile psychoacoustic effect. [While I have continued some work in this area, I have not found a consistent, definitive cause/effect relationship. However, it is clear to me that the audibility of acoustic polarity inversion is dependent on both the acuity of the listener and the nature of the program material, and not highly dependent on the transducers involved.]

4. The issue of the audibility of acoustic polarity inversion is not a matter of black and white but of a series of shades of gray, seemingly dependent upon the simplicity or complexity of the program material being auditioned, to some small extent upon similar factors of complexity of the listening environment, and to some extent upon the sensitivity of the listener. [While I would like to see standardization of polarity in recording and reproduction, it seems to be a minor issue compared to others that affect sound reproduction much more strongly.]

Several final issues need to be laid to rest. One is the question of the use of digital recordings and digital program material to do listening experiments. The essential results described above have been duplicated with real instruments, microphones, and headphones in real time without the use of any recording devices. Some fanatics suggest that only they can hear things because of their equipment. This is total nonsense.

Some have suggested that the quality of the loudspeaker and/or measurement techniques used for the original paper were somehow defective. (This has been implied by some snide remarks by C. Johnsen in *The Audio Critic* letters column as well as in *Audio* magazine.) Our experiments were set up with extreme care, using a large array of the very best professional-level instrumentation equipment in my Electroacoustics Laboratory. We are totally comfortable that we know how to use this equipment, have designed a suitable loudspeaker, and have analyzed the results in an entirely professional manner.

Finally, all of our findings have been confirmed with headphones of various sorts and a number of quite diversely designed loudspeakers. The design of the headphones or loudspeakers, within reason, is in my experience irrelevant to revealing the phenomenon.

I have over the years discussed these issues with many AES members, including Richard Heyser and Stan Lipshitz. I believe that we all would prefer that the industry took care with polarity conventions. But, they have not for the most part. Polarity is simply not a high-priority issue for most professionals and certainly not highly important for the enjoyment of reproduced sound.

Nevertheless, I am actively carrying out additional experiments and I hope to pursue the matter with the goal of finding sources/causes of audibility of acoustic polarity inversion and to specify it more clearly in a scientific, responsible manner.

That's it for the time being. I suppose that heated debate will continue, just as it does with the cable/interconnect issue.

Very sincerely,
R. A. Greiner
Professor
Fellow of the AES

*Thank you for the kind words about **The Audio Critic**. We not only get "some very good people" to write articles for us but also, as your example proves, some very good people to write letters to the Editor. Indeed, your letter dots the i's and crosses the t's on the subject of polarity for all rational audiophiles. Let the tweaks read it and weep.*

—Ed.

The Audio Critic:

I love you curmudgeons. You're even occasionally correct—but then, so are John [Atkinson] and Harry [Pearson].

I have no doubts as to your superior theoretical and technical qualifications; I also have no doubt that much subjective reviewing necessarily utilizes poor methodology; however, why do *you* have problems accepting the idea that some audible differences are either difficult to measure using conventional parameters, or are the result of phenomena not yet fully understood?

Howard Cowan
Woodland Hills, CA

I have no problem whatsoever with the ideas you state as long as those "audible differences" are indeed audible. What I have a problem with is the statement that "I can hear the difference" when you are unable to prove to me under controlled conditions that you can actually hear it. If there really is a provably audible difference, the cause may or may not be easy to determine—that's a totally separate issue.

As for John and Harry, see "The Doctor Zaius Syndrome" (page 10).

—Ed.

The Audio Critic:

I have some ideas about how I think a modern music-reproducing system ought to be be configured to minimize hardware interactions with the music information. I think from what I've read in this magazine, and from some components already placed on the market (Meridian comes to mind), that some very talented people are thinking along these same lines, and I'm wondering why we (meaning the industry and the hobbyists) are not headed a bit more quickly in this direction.

Before I explain, let me admit (as you're wont to question this) that I have no credentials other than a 30-some-year interest in the hobby. I'm also a fairly recent convert from the music-to-justify-hardware group.

My concept of a system would have it divided into two basic modules. One I'll call the control module, the other the speaker-system module—or actually modules, as there would be several of these. The control module would look very much like a home computer system or possibly a TV set, and might actually be integrated with one of these—or both. The purely electronic functions, such as preamp, tuner, processor, etc., would be installed as industry-standard plug-in boards, with all of their switching and control functions accessed as icons on the screen—very similar to Macintosh, or IBM with Windows, menus. Access to the sound system might actually be a menu option on your computer terminal. The actual hands-on control would be a mouse or infrared remote.

The control module would provide ports with a standard multipin socket, for inputs from purely mechanical program sources such as an LP turntable *[buggy whip on the space shuttle?—Ed]*, CD transport, cassette transport, etc. The module would operate entirely in the digital domain, which would eliminate the need for expensive hardware to maintain a clean signal, and the output, or outputs, to the speaker systems would be via fiber-optic cables. System upgrades would be accomplished by replacing or adding boards.

The speaker systems would each contain a speaker, or speakers, a bandwidth-limited amp with equalization (if necessary), an electronic crossover, and a DAC circuit for the fiber-optic inputs. These systems would be tailored for their particular function, such as subwoofer (similar to the Velodynes), main left and right, and surround-type speakers. I visualize the surround speakers as designed much like track lighting on the ceiling, spherical enclosures that would be aimable and could contain their electronics in small boxes which could flush-mount in the wall or ceiling and be attached by short cables to the spheres. A full system would consist of two subwoofers limited to 80 Hz, two mains for left and right (80 Hz and up), and four of the ceiling-mounted speakers, center front and back, and back left and right.

Now tell me, where am I wrong in this concept? And, if it's basically accurate, why aren't we already there? Disregarding the tubes-and-LP crowd, I think we're hung up in oldthink. I, for one, find the stack of black chassis and their tangle of wires an eyesore and probably unnecessary.

Hartley Anderson
Waco, TX

To quote that song from the big-band era, "I'll Buy That Dream." As you point out, bits and pieces of the dream exist already: powered subwoofers are the rule rather than the exception; powered full-range speakers are still the exception but no longer a great rarity; Marantz showed a computer front end as early as 1991; I could go on. It hasn't all come together, though; the demand isn't there; "separates" are still the audiophile norm.

Your basic concept is very much in line with my own thinking, but I'll go even further: A/D conversion should take place right out of the microphone preamp and the signal kept in the digital domain throughout the recording, editing, mastering, duplicating, broadcasting, domestic playback, etc., processes, right up to the D/A conversion just before the amplification stage of each separately powered speaker channel—and that includes digital filters for all crossovers. (Maybe you, too, had that in mind but didn 't quite say it.) The speaker deployment should probably follow the Lexicon model in its fullest form: front left, center, and rear, subwoofer(s), side left and right, rear left and right. But these are details. You've got the main idea right—and you will see it happen. The question is, when? Some observers feel that the audio consumer will continue to resist the idea of the separation of amplifiers and speakers.

—Ed.

The Audio Critic:

...My question specifically has to do with whether a pair of subwoofers is better than one subwoofer. This question is generated by a recent article by John F. Sehring, which appeared in *Audio* magazine (February 1994). In that article, Mr. Sehring gives a *number* of reasons why stereo subwoofing is superior. I wonder whether *The Audio Critic* has an opinion in this regard. I also realize that the answer might depend partly upon a number of variables (placement, room size, room acoustics, etc.), and therefore no universal answer or rules of thumb may obtain. However, any opinion at all would be helpful.

I also have another question regarding the use of built-in amplifiers in subwoofers. Some recent literature which I received from VMPS suggested that built-in amplifiers are a bad idea because subwoofer vibrations will eventually simply rattle them apart, as it were. Does this turn out to be the case? Do the electronics in powered subwoofers self-destruct after a relatively short life span?

Please keep up the good work; your magazine is a delight.

Sincerely,
David R. Reich
Auburn, NY

/ have always been of the opinion that a pair of stereo subwoofers is preferable to a single mono (L + R matrixed) subwoofer—see Issue No. 16, page 16—but Tom Nousaine, who has studied the subject in considerable depth, vigorously disagrees. His findings are documented in a forthcoming article in the January 1995 issue of Stereo Review. *This looks like one of the few legitimate controversies in audio (unlike the nonsense about blind tests, tubes, etc.), and I am quite open to all arguments. But, as in other debates about all but the most obvious audio phenomena, a number of reliable practitioners have to be able to repeat the same tests and obtain the same results. Maybe Tom Nousaine needs to broaden his statistical base before coming to a sweeping conclusion; maybe not. (For one thing, he is not into classical music; as I once told him, it's a case of "Pop Goes the Weasel.")*

The VMPS caveat sounds like sour grapes to me, since they make and sell only passive subwoofers. Why don't built-in crossover networks, whose large components and large boards are much more prone to vibration than amplifier parts, fall apart untimely? Why don't radios in jeeps fall apart? Needless to say, a certain amount of care and competence in construction and placement must be assumed in all such instances. Audio hypochondria is, of course, a proven marketing platform.

—Ed.

The Audio Critic:

...I...noted that David Rich made reference to the Marantz CD-63 in his footnote on page 16 of Issue No. 21. However, I was disappointed to see that he (Editor?) refers to the unit as "Philips-designed." Ordinarily, such a reference would be taken as a compliment. Indeed, in the *Journal of the Audio Engineering Society,* I noted some years ago in their regular review of audio patents that the reviewer referred to Philips in the following way: "In many ways, Philips is the audio equivalent of Mercedes-Benz. If there is an esoteric way of doing things, this is how Philips will do them." As I seem to have misplaced the particular issue, I cannot say that the above is an exact quote, but it surely is very, very close to the original comment.

In the case of the Marantz CD-63, the model is in fact wholly designed within the Marantz organization. Specifically, our chief CD designer, Mr. Yoshiyuki Tanaka, is the gentleman responsible for the CD-63 and most of our other CD players. He has a strong technical background in digital audio as well as analog circuit design, including power supplies, and is very familiar with a wide variety of available devices, CD mecha nisms, and the like. I have in the past forwarded to him articles of interest, most of which have come from the pages of your magazine.

I am always pleased to see mention of our gear in your magazine, and at Marantz we are always quite proud of the Marantz association within the giant Philips concern, but I would ask that in the future products submitted for review such as the CD-63 be referred to as Marantz-designed, as this is the reality. Philips per se had nothing to do with the CD-63 design; however, we happily acknowledge the fact that the CD-63 employs a number of Philips-originated components, such as the CDM12 mechanism, TDA1301 servo, SAA7345 decoder, etc., that are also found in Philips-branded models, as well as models from other firms, such as Audio Research....

As always, I appreciate your interest in Marantz gear.

Best regards,
David Birch-Jones
Marketing Manager
Marantz America, Inc.
Roselle, IL

You'11 find that the review of the Marantz CD-63/63SE in this issue, essentially an updated leftover from Issue No. 21, has been annotated to reflect your input. That purely Marantz, non-Philips engineering seems to come up with products that offer very solid performance per dollar. As for the articles you send to Mr. Tanaka, I can understand why they are from this publication, not from the digital cloud-cuckoo-land of...well, you know.

—Ed.

The Audio Critic:

Having just completed reading the review of the Magneplanar MG-1.5/QR, I believe some important issues are raised. Audiophiles know well that assessing loudspeaker performance is difficult because it is to a large extent subjective. Every loudspeaker design is a compromise, and no particular transducer technology has exclusive rights to accuracy. Given your statements at the outset of the review, some would question your "detached objectivity" in this case. It is clear that you hold planar magnetic loudspeaker technology in low regard and this appears to have predestined your conclusions.

One of the generic criticisms of Magneplanar designs made in the review is the lack of low bass capacity. Your nearfield measurements of the MG-1.5/QR indicate its response extends to approximately 40 Hz. The farfield response in my listening room is significantly flatter than you suggest (±4 dB), and there is usable bass down to approximately 30 Hz. Static distortion in the bass is low (did you even bother to measure it?). The conclusion that the MG-1.5/QR has "not enough bass" doesn't follow, particularly when compared to other loudspeakers available at the same price.

The other major criticism presented involves driver ringing. This has been noted by others in the past and appears to be related to the physical nature of the driver. It has been described in a variety of uniformly driven nonrigid drivers, which include ribbon and electrostatic systems in addition to Magneplanar drivers. The audibility of the effect has not been established, to my knowledge, as it relates to a farfield listening position. Transducers with significant energy storage typically emphasize and smear consonants in the spoken voice and have poor square wave response, neither of which is true of this speaker.

As for the "not enough focus, too much coloration" that is reported in the review, I can only comment that in my experience the MG-1.5/QR is an extremely revealing transducer that allows a wealth of musical detail to emerge. A common characteristic of neutral transducers is that each recording played sounds different. This is exactly how transparent the MG-1.5/QR sounds. Perhaps the Editor heard what he thought he measured rather than the reverse.

None of the many conventional loudspeakers that I am aware of in its price range produces as wide, deep or stable an image as the MG-1.5/QR. These aspects of performance are not even discussed in the review. The LEDR tracks from the first Chesky test disc provide an effective and quick method of evaluating the spacial characteristics of loudspeakers in conjunction with the surrounding environment. Few loudspeakers can generate substantial "height" with the vertical signal on the disc or provide an apparent soundstage wider than the stereo pair. With the MG-1.5/QR, the signal extends to the ceiling in the vertical test and beyond the width of the stereo pair in the horizontal test.

The rather cursory nature of your loudpeaker reviews has concerned me since I first subcribed to your journal. Perhaps it is time for the Editor to apply the same rigorous standards to loudspeaker reviews as Dr. Rich does with the various electronic components he discusses. May I suggest that in the future more of the test results and discussion of the listening conditions (especially speaker placement) be provided. It would also be useful to have the manufacturers respond to issues raised in the reviews. These changes would allow the reader to better assess the relative merits of each design and reach his own conclusions.

Yours truly,
Dr. Douglas M. Hughes
Rochester, MN

I'm not surprised that the Minnesota audio mafia finds staunch supporters at the Mayo Clinic (or am I misinterpreting your prefix and your address?), but you happen to be mistaken on most of the points you bring up. Not all of them, though.

You 're right when you say that loudspeaker evaluation is highly subjective, although I try to back up my subjective opinions with objectively verifiable evidence. Furthermore, my subjectivity has been refined over the years through exposure to literally hundreds of speakers—/ have some very good reference points of subjective comparison.

You're also right in observing that my loudspeaker reviews are not quite as rigorous as David Rich's reviews of electronic components, but there are good reasons for that. A typical electronic signal path in audio (such as, say, the left channel of a power amplifier) has one input and one output. It's relatively simple and straightforward to examine the I/O relationship. A speaker, on the other hand, has one input and n *outputs. Which of the latter do we examine? Where in space is the valid, or "official," output of a loudspeaker system? How many points in space are sufficient to give us an accurate picture of the total output? These are nagging questions of measurement methodology, and then there are the endless other questions such as the physical difficulty of ABX comparisons of speakers (see Issue No. 20, page 39) and the various biases introduced by the reviewer's accustomed listening room, etc. Rigorous standards? We're working on them. Don Keele is perhaps the most rigorous loudspeaker reviewer of us all, but then I do a few tests that he doesn't. Amplifier-like certainty in speaker testing we don't have—and will not have soon.*

Now then, here are the points where you are wrong. (1) I do not "hold planar *magnetic loudspeaker technology in low regard,"* au contraire, *my MG-1.5/QR review begins with "Yes, I have a soft spot," etc., and later I enthuse over the upper bass and lower midrange of the Tympani IVa. (2) The lower bass of the MG-1.5/QR is defined by the fundamental resonance of 44 Hz; it may be that* your *particular room happens to offer some reinforcement down to 30 Hz, but the designer can't count on that in* my *particular room. (3) My frequency response measurement was anechoic (via MLS) and therefore closer to reality than any in-room measurement. I don't know how you arrived at your ±4 dB in-room figure, but it's irrelevant; no speaker designer deliberately makes the anechoic response nice and jagged in the hope that the room will homogenize it. (4) The (true) ribbon tweeter of the Tympani IVa doesn't ring, so your generalization is incorrect. (5) The audible effects of ringing depend on the frequency but they are very real; having been deeply involved in speaker design as well as testing, I can only say I wish you were right. (6) The MG-1.5/QR* does *have poor square-wave response; furthermore, ringing doesn't necessarily affect the square-wave response unless located near the square-wave fundamental or its odd harmonics. (7) Revealingness is a relative quality—revealing compared to what other speaker? (8) I listen* before *I measure. (9) I* did *comment on the excellent height and width of the soundstage. (10) Just as a single example, the ACI (Audio Concepts, Inc.) G3 speaker, reviewed in Issue No. 19, beats the MG-1.5/QR on bass and just about everything else, at less than two thirds the price.*

As for manufacturers' comments, we publish every word of them unedited—if they write us. I don't believe in letting them preview the reviews, however; it makes for fruitless preemptive hassles.

Final thought: your Magneplanar MG-1.5/QR is every bit as good, or bad, after my review as it was before it. You are quite certain that it's a great speaker, so why is it important to you that it should be blessed by ***The Audio Critic*** *?*

—Ed.

The Audio Critic:

Most, if not all, of the articles appearing in *The Audio Critic* are informative, so I would like to suggest that some knowledgeable individual write a short article on what qualities make something sound live. When one walks into a dining room or bar and hears a piano, it is easy to tell with the first chord if the music is live or is being reproduced, and sometimes even the make of the piano. It is one of the reasons I have stopped going to most concerts, as they seem to feel they must use speakers, and therefore I'm forced to listen to a loudspeaker (not necessarily a good speaker) and not the instrument itself. If I go to a concert I want to hear the actual horn or piano or whatever. If I want to hear speakers, I can stay home, as I've better speakers than they use and far better sound.

I happen to be a collector of jazz from the 1920s to the 1950s and still buy 78 rpm records when I find something in very good condition that I want. I collect these old records because many of these recordings have not found their way onto LPs or CDs. The bonus is that the music on these 78 rpm records has more of a live sound than LPs and much more than CDs. I think Doug Sax touched on this once. I truly believe, and hope, that I'm not being influenced by the scratch and noise common to these old records.

My cassette deck, a Nakamichi 680ZX, has a plasma display in place of a VU meter or LEDs as level indicators. When I copy a CD or even a 33-rpm LP record onto a cassette, it is easy to see the lighted portion of the display move from left to right on drum rim shots or struck piano notes (anything percussive). By that, I mean one can easily see the lighted portion travel from say -40 to 0 dB. Although fast, it's easy to see it move up the scale with an increased level. When I copy a 78-rpm record onto a cassette, it's different. The music going from -40 dB to 0 dB will result in the entire display, up to 0 dB, instantly being lit. It is so fast that one cannot see any movement of the lighted display; it just appears. This instant change in level also happens if I record the grand piano in our great room. All of this is not as noticeable on LED displays and completely obscured on VU meters, as both are slow in comparison to respond. Maybe that is why there are so few plasma displays.

This leads me to believe that the attack time of a sound seems to have a direct bearing on how live it appears, and a live sound is what we are all striving for. Even when played through a 5 or 8 kHz lowpass filter, 78-rpm records have a live sound. To focus my question, why doesn't a CD sound as live as a 78-rpm record? By eliminating the compressors and limiters and *minimizing the active stages and feedback,* surely we should be able to archive CDs that sound as live as the old 78-rpm records of sixty years ago. The added benefit of having no noise would be wonderful.

I have every issue of *The Audio Critic,* having read all of them at least twice. Keep up the good work, as there is so much #*%!# being shoveled out there....

Yours truly,
Thomas F. Burroughs
Prescott, AZ

Aren't you the Tom Burroughs who was selling Klipschorns in New York City circa 1951? (I was very, very young then, of course, and so were you.) If so, here's some advice from one geezer to another:

Finagle yourself an invitation to a live recording session (of a reasonably competent label, I should add). Listen to the direct sound of the musicians. That's live, right? Now step into the monitor room. (I'm assuming something a little better than a telephone booth, and decent monitor speakers.) What you now hear is the CD sound (via the same l's and 0's as will appear on the CD). How "live " is it? Do you now feel that 78-rpm shellac would sound more nearly like the live musicians? I seriously doubt that you would feel that after such an exercise. I think you have lapsed into some kind of technostalgia (to coin a word).

Of course, a superb 78-rpm recording from 1947, pressed on vinyl (which they started to use around then) may sound more "live" than an indifferent LP from 1951, which in turn may sound better than a botched CD from 1984 (they had a few of those). But the best 78s versus the best LPs versus the best CDs? Get out!

I have no idea what's with your level indicator—it could be any number of things, including overload—but it isn't attack time you're measuring. The leading edge of a dynamic peak is determined by the high frequencies, and the 78-rpm shellac medium was certainly not superior in bandwidth to LP and CD. As for compression, yes, it can upset the applecart, but good CDs aren't compressed.

And, by the way, feedback (correctly applied feedback) is not the bad guy. That's a 1970s notion, meanwhile laid to rest by some of the best minds in the engineering world. Gotta keep up with the *times, old-timer. (Oops, what if you're not* that *Tom Burroughs? What a burn...)*

—Ed.

The Audio Critic:

I am a recent subscriber and I am unbelievably happy to have found *The Audio Critic.* I assembled a good system for the first time in the last year, so I assume that I am the kind of person that other magazines whine about needing to recruit to save the "High End." Please allow me to give you my perspective as an inquisitive novice.

After reading enough analog drivel every month, I actually started to wonder if I had made a mistake by selling my LP collection 4 years ago, so I went down to my favorite hi-fi dealer and compared a CD player with a much higher-priced turntable/cartridge combo. The CD front end was so vastly superior in detail, dynamics, noise, and overall quality that I remembered instantly why I smiled the first time I heard a good CD system. I also thought it was pretty darn *musical* too, whatever that may be.

I am constantly bombarded with articles recommending unbelievably expensive wires, interconnects and, most recently, magic wooden disks. I can't hear a difference in controlled blind tests and neither can the people that recommend them, but if you don't agree with them you are some kind of uncultured ignoramus. Isn't it convenient that the differences are supposed to be unmeasurable things such as *dynamic bloom* and *liquidity.*

I guess some people like distortion in their music and that's why they love those outrageously expensive vacuum-tube amps. I think that the real reason is that a lot of people went over to Grandpa's house as a kid, and his stereo glowed in the dark. Now that they have six-figure incomes they'll be damned if theirs isn't going to glow too! It's a good thing we have those East Bloc 1930s economies to supply us with 1930s-technology vacuum tubes.

When I entered the High End I had no inkling that it was a fantasy world inhabited by mystics, romantics, and charlatans. The High End is hurting and attracts almost no women (another frequent lament) because it is dominated by reviewers, retailers, and manufacturers that are either greedy or foolish and have lost touch with reality. I don't think a lot of them realize how bizarre it looks from the outside.

The Audio Critic is like a breath of fresh air. I'm glad that logic, reason, and facts have a proponent in the audio world. Keep up the good work!

Darren Leite
Scottsdale, AZ

Everything you say is right on the money, but you don't ask the sixty-four-dollar question:

Is the truth bad for business?

My answer is that, in the high-end audio world, the truth is probably bad for business this week and next month but very good for business over the next ten years. B.S. has a limited shelf life; given sufficient time, most consumers tend to switch to the truth. The tweako/weirdo high-end promoters, however, don't think that far ahead.

Thank you for your kind words.

—Ed.

The Audio Critic:

I would like to know if you have an explanation for the fact that, in my system, the Parasound HCA-2200II sounds noticeably better through its balanced inputs. I have a Parasound P/LD-1500 driving it, with 24' lengths of Straight Wire Flexconnect (unbalanced) or Canare Star-Quad (blanced). The levels are matched to within about 0.2 dB, and out of 10 trials, single-blind (I was unaware of the choices, but the switcher was), the difference was immediately apparent each time.

Yours truly,
Rob Bertrando
Reno, NV

As David Rich's review in Issue No. 21 clearly explained, the simplistic input buffer circuit makes the balanced-input distortion of the Parasound HCA-2200II more than an order of magnitude worse than through the unbalanced input. The distortion is probably still below the threshold of audibility. Parasound has acknowledged that there was some kind of minor foul-up in the production version of the balanced input circuit, and we were supposed to get a letter from John Curl that would clarify the matter. We are still waiting.

"Sounds noticeably better" is of course a purely subjective opinion and unprovable. "The difference was immediately apparent" is, on the other hand, a provable statement and probably true in your case.

Here are the possibilities that I see: (1) The balanced-input distortion was just above the threshold of audibility in your unit, and you liked it. (2) The level matching wasn't good enough ("about" 0.2 dB could have been 0.3 dB, which is often perceptible), and you liked the louder or the softer choice. (3) The tweako Straight Wire interconnect, which I'm not familiar with, may introduce a rolloff or other marginally audible inaccuracy, and you liked it.

One thing is certain: the I/O relationship is more linear when the unbalanced input of the HCA-2200II is used.

—Ed.

The Audio Critic:

I am writing to respond to Mark S. Willliamson's letter, as well as the excellent article on clock jitter by Robert W. Adams, in Issue No. 21. First, Mr. Williamson states, "My only regret is that some of the information is a little technical for those 'laymen' who are not part of the engineering kingdom. I would be grateful if you could dilute some of the techno-lingo from time to time."

Williamson has been complementary to *The Audio Critic* as am I; however, the idea of catering to a less technical (or nontechnical) readership sends shivers up my spine. Dilute? To what end? I'm not trying to attack this man. I just don't want you to do what he says. I have been reading audio publications for 25 years, with a craving for the detailed articles and technical subjects that appear in every issue of *The Audio Critic.* There are many audio publications that are already diluted for those who have no stomach for details. If I want serious, scientific, technical analysis of audio issues, there are only two choices: *TAC* at $24 per year, or pay hundreds for a professional journal such as that of the Audio Engineering Society (AES). I worry that there will be continuous pressure on *TAC* to broaden its appeal by deleting the important details and writing for the marketers. I must admit to you that I am a practicing electronics engineer with a B.S.E.E. under my belt, so it figures that the technical details *would* be much to my liking. I do not wish to demean Mr. Williamson's comments at all, and am pleased that he has joined the thinking among us. The physics of the universe is a complicated thing, and we as mortals must use difficult technical language to describe it with any accuracy at all.

Robert W. Adams's article on clock jitter was simply outstanding, providing a great analysis of the specifics of clock-jitter effects on digital-to-analog conversion. This is a good example of what I am talking about. Mr. Adams is in a position (as few are) to write this article. We need just these kinds of articles in order to really understand what is going on, and what is important. David Rich has written several detailed articles that spoke to me as a designer and engineer, sending me scurrying for my semiconductor data books and reviewing circuit theory. Excellent. I have been waiting many years for this kind of reading from the audio press. Reading *TAC* is actually challenging for me. I get through a *Stereo Review* in about 15 minutes, but *TAC* takes weeks to fully absorb.

Diluted summary: If you change anything I'm going to get really mad.

Clark Oden
Project Engineer
Frontier Engineering, Inc.
Oklahoma City, OK

Just how technical should a responsible audiophile journal be? It's a difficult question. If we oversimplify, we become superficial—and there are already plenty of other superficial audio magazines. If we speak mainly to the E.E. sensibility, we lose the vast majority of our readers. We must strike a delicate balance. As I've stated before, The Audio Critic is not "My First Book of Electricity." If you don't know what impedance is, or what a FET is, you'11 have to find out elsewhere. On the other hand, we aren't the "Journal of the AES, Junior," either. We try to keep the math to an absolute minimum (although David Rich always wants to sneak in more and more). I want the reader who doesn't understand, say, 20% of what we publish to understand the remaining 80% perfectly. That way the 100% understanding can be expected to come eventually. Our basic conclusions must be crystal clear to everyone—and I think they are.

One thing we can do is to break out the highly technical stuff in sidebars, so it doesn't slow down the nontechnical reader of the main article. Sometimes the article is written in such a way that it's very difficult separate the rough from the smooth, but as you know we try to do it when we can.

Don't worry, we don't intend to "dilute " our technical accountability. And I agree completely with your beautiful sentence about the physics of the universe.

—Ed

The Audio Critic:

.. .Keep up the tweak bashing!

Seriously, I have been working on two processes relating to digital audio recently—sample-rate conversion and noise-shaped dithering. In an effort to find out what is really bearable vs. what is measurable, I became sucked into the audiophile-tweak world—I was convinced my sound system wasn't good enough because I couldn't hear characteristics I could measure. After all, these golden ears doing reviews can obviously hear the differences—right?

I found your magazine in time to be rescued from a fate worse than death—financial and otherwise. Thanks.

Regards,
Bruce Hemingway
Hemingway Consulting
dB Technologies, Inc.
Seattle, WA

It's useful to know what to say first, right off the bat, to those golden ears who claim to hear the differences you can't. You say, "No, you can't hear that. You're just telling me you can but you'll never be able to prove it." That immediately steers the discussion in the right direction without allowing it to go off on some highfalutin, abstract, pseudoscientific, psychobabble tangent—which is what I find worse than death.

—Ed.

The Audio Critic:

To the little person "in the smallest room of [his] house" [Issue No. 21, page 4]. The Editor gave you credit for writing, but we know it was copying, don't we? You forgot the quotation marks on "*/ am...* and *...behind me."* Then you forgot to give the composer Max Reger credit for the quote.

You could have signed your name. We would have understood the X.

Ron Garber
La Porte, IN

What an erudite subscriber! What a sucker of an Editor! What a scummy anonymous letter writer!

I looked it up, and you're absolutely right of course. Here is what Max Reger wrote to the Munich critic Rudolph Louis in response to the latter's review in the Münchener Neueste Nachrichten, *February 7, 1906:*

Ich sitze in dem kleinsten Zimmer in meinem Hause. Ich habe Ihre Kritik vor mir. Im nächsten Augenblick wird sie hinter mir sein.

Translation: "I am sitting in the smallest room in my house. I have your review before me. In a moment it will be behind me."

One must understand that in Europe in 1906 the use of newspaper for hygienic purposes was quite common. As for the quip, Max Reger (1) obviously made it up himself and (2) signed his name.

—Ed.

Coming:

➡ A review in depth of the $19,000 Snell Acoustics Type A loudspeaker system, along with other interesting speakers.
➡ Reviews of high-quality surround-sound processors and preamplifiers, from Lexicon, Marantz, B&K Components, and others.
➡ The long-promised survey of FM tuners and indoor antennas (really!).
➡ Still more reviews of power amplifiers and preamplifiers (they keep coming).
➡ Further evaluation of perceptual coding technologies and hardware.
➡ Some big surprises (you'll never guess).

Paradoxes and Ironies of the Audio World:

The Doctor Zaius Syndrome

By Peter Aczel
Editor and Publisher

When the truth is so terrible that admitting it would surely make the whole system crumble, ape logic demands denial and coverup.

Have you ever seen that marvelous 1967 science-fiction movie *The Planet of the Apes?* If you have, you will recall that it depicts a planet of the future where English-speaking anthropoid apes are the rulers and humans are speechless beasts of burden, enslaved by the apes and despised as a totally inferior species. The apes have horses and guns but no real technology. Doctor Zaius, the subtle and highly articulate orangutan who is this society's "Minister of Science and Defender of the Faith" (he is played by the great Maurice Evans), knows something the other apes do not: that humans in a past era possessed not only speech but superior technology, flying machines, powerful weapons, and so forth, all of which served only to bring about their eventual downfall and reduce them to their present condition. Doctor Zaius fervently believes that any knowledge of this truth about humans would totally destabilize the society of apes and result in the end of their world. The ape dogma he fanatically protects, even though he knows better, is a blatant denial and coverup of the actual history of the vanished human civilization and a paean to the eternal superiority of the ape.

I won't give away the rest of the plot to those of our readers who haven't seen the movie and may want to, but doesn't Doctor Zaius resemble certain key figures in the high-end audio community? He knows the truth but it's bad for the establishment. The system would come crashing down if the truth were revealed. To pick an obvious example, consider John Atkinson, the subtle and highly articulate editor of *Stereophile.* Don't you think he knows? Of course he knows. But if he admitted that $3000-a-pair speaker cable is a shameless rip-off or that a $7000 amplifier sounds no different from a $1400 one, the edifice of high-end audio would begin to totter—or so he thinks (and may quite possibly be right). Consequently, he spouts convoluted scriptural arguments and epistemological sophistries, just like Doctor Zaius, in order to pervert the obvious, uncomplicated, devastating truth.

There is a perfect illustration of this process in the August 1994 issue of *Stereophile,* where Zaius-Atkinson once again bashes blind listening tests in an "As We See It" editorial. Such tests are of course considered extremely threatening by a publication that reports night-and-day differences in sound which absolutely nobody can hear when the levels are matched and the brand names concealed. He brings up all kinds of intricate flaws and drawbacks that may very well exist in *some* blind tests but turns his back on the large number of blind tests in which all of his objections have been anticipated and eliminated and which nevertheless yield a no-difference result every time. He knows very well, for example, that no one has ever, ever proved a consistently audible difference between two amplifiers having high input impedance, low output impedance, and low distortion, when operated at matched levels and not clipped—but like Doctor Zaius he conceals that knowledge. He'd rather collect rare case histories of screwed-up blind tests than deal with the vast body of correctly managed blind tests that undermine the *Stereophile* agenda. (Just for the record, I'll state for the nth time that there are only two unbreakable rules in blind testing: matched levels and no peeking at the nameplates. To eliminate "stress," take a week or a month for each test, send everybody else out of the room, operate the switch yourself at all times, switch only twice a day—whatever. The results will still be the same.)

A hard-nosed insight by the Weasel.

Our columnist Tom Nousaine (a.k.a. the Weasel), in a recent conversation with me, stated his belief that any longtime audio reviewer who has tested hundreds of different audio components over the years knows exactly what the truth is about soundalikes because it is utterly impossible to escape that truth after so much hands-on experience. It asserts itself loud and clear, again and again. Therefore, he argued, the audio journalists who invariably report important sonic differences are most likely a bunch of hypocrites, i.e., exhibit the Doctor Zaius Syndrome. I was strongly inclined to agree with him, but then I said, "Well, what about Bob Harley?" We agreed that Harley could be an exception. He may very well be sincere because he just doesn't get it, not even after all these years. Larry Archibald, on the other hand, is smart and tough and definitely knows the truth, we felt. He is probably the biggest Doctor Zaius of them all, ready to make a monkey of any insufficiently enlightened audiophile. At the risk of offending against the principle of *de mortuis nil nisi bonum,* I'm willing to venture the opinion that even the late Bert Whyte and Len Feldman, regardless of their other important contributions, did a Doctor Zaius number on certain audio issues rather than face the wrath of the tweaks and the accusation of heresy. As for Harry Pearson and company, who knows? Are astrologers, shamans, and witch doctors sincere or hypocritical? As long as they don't try to usurp scientific arguments, what difference does it make? And if they do try, they're pathetically ineffective anyway.

A serious credibility gap.

In the same issue of *Stereophile* as the John Atkinson blind-test-bashing editorial, Larry Archibald views with alarm the low bit-rate coding scene in an open letter to Pioneer. He wants them to hold off on the implementation of the Dolby AC-3 coding standard for LaserDisc because it may not be the highest-quality solution sonically. In other words, he suddenly doffs his orangutan suit and shows concern for something that may actually be true, i.e., audible.

Well, you blew it, Larry baby. You went ape—or cried wolf, to mix my animal metaphors—so many times about low-credibility tweako matters that on the Pioneer level of big-money decision making you are no longer taken seriously even when you may have a perfectly legitimate, nontweako argument. That's quite obvious from the two replies by Pioneer executives printed in the September issue, both of which basically tell you to relax, tweak boy, take it easy, and let the real experts get on with their work—at least that's the way I read them. You may conceivably end up being right, and Pioneer wrong, about Dolby AC-3, but you're clearly wasting your breath. (See also Tom Nousaine's column in this issue for a somewhat different perspective.)

By the way...

As I recently noted with a poignant sense of recognition, *Stereophile's* visual leitmotiv for blind testing is the familiar three apes with hands on eyes (get it?), ears, and mouth. Now you know why. It isn't just an art director's passing fancy. Doctor Zaius can feel right at home.

Why do I even bother to tell you all this?

All of our readers who have been with us for more than just one or two issues are aware of my enormous frustration on the subject of scientific truth in audio. The very idea of a Doctor Zaius Syndrome, even it's only a parody, suggests the existence of antiscience in audio as a *tradition,* not just a momentary aberration—and a tradition it is, going back to the early 1970s, at the very least. In the late '40s and throughout the '50s and '60s, whatever the most highly qualified and experienced engineers said about audio was the accepted truth. Then came post-modern irrationalism, post-Watergate anomie, fortune tellers in high places, pyramid power, Jesus-haired record-store clerks as self-proclaimed audio experts, untutored high-end journals, pooh-poohing of engineering societies, derision of degreed academics—the B.S. era of audio (and I don't mean Bachelor of Science). Today, the melancholy truth is that tweako cultism has become mainstream audio, at least above a certain price range, and engineering facts are regarded as disturbingly radical or at least eccentric. The scientific audio community has been marginalized.

I despair at this point of a journalistic solution. Even if *The Audio Critic* increased its circulation by a factor of 50 overnight—I'm being deliberately absurd—it might still be too late for the message. The cultists have been too deeply indoctrinated and too long. The pimply-faced kid in the Bon Jovi T-shirt who tried to sell you AudioQuest Sorbothane Feet (the bigger kind) in your local audio salon is not going to change his belief system. Not in this antirationalist age and culture.

I can think of only one effective remedy. Many years ago, long before our younger readers became interested in audio, the Federal Trade Commission put an end to fraudulent power-output claims in amplifiers. Today, the power-output specification must take the form of "200 watts rms into 8 ohms from 20 Hz to 20 kHz at less than 0.25% total harmonic distortion." Before then, the same amplifier could have claimed 800 watts because it could produce that for 2 milliseconds at 1 kHz into 2 ohms with 10% distortion. What if the FTC suddenly became interested in audio cable advertising, for example? That chattering sound you hear comes from the teeth of cable vendors at the mere mention of the possibility. And that low, rumbling sound you hear is Doctor Zaius growling, "That's heresy!"

Anyone out there whose nephew or brother-in-law is a young, crusading, Ralph-Nader-like employee of the FTC? Get him interested!

* * *

P.S. Long after the above was written, just before press time, I received a PR release from one of *Stereophile's* flacks, hyping the magazine's willingness to tell the "absolute truth" about a product even at the risk of losing the advertising support of the manufacturer. The latest editorial by Larry Archibald-Zaius simultaneously proclaims the same lofty principle. The Velodyne brouhaha is used in both instances as proof: they panned the DF-661 speaker; Velodyne canceled all its ads; see how incorruptible they are. Hey, you can't buy off the Defenders of the High End Faith with a few ads when they face the deadly threat of a midpriced super speaker!

"I [don't] see any reason...why a magazine couldn't have both principles and commercial success," the PR release quotes Archibald-Zaius. "I've never had even a second thought on the subject."

The monkey doth protest too much, methinks. •

Loudspeakers Are Getting Better and Better

By Peter Aczel
Editor and Publisher

The proof is in the recent designs that nudge the state of the art, particularly in bass reproduction, but also in minimizing distortion over the full range.

I can't say it often enough: if you already own a fairly decent home music system, nothing can significantly change the quality of your audio life except new and better loudspeakers. They are *so* much more important than preamps, power amps, CD players, etc. What happens in most audiophile households, however, is that the main speakers are firmly embedded in the living-room décor, so that any major change is subject to vigorous spousal objection, especially if larger speakers are contemplated. The typical audiophile then satisfies his lust for shiny new equipment by buying, say, a new preamplifier and persuading himself that the sound is now much better, when in effect it hasn't changed the least bit. It's a syndrome that depresses the hell out of me.

(Incidentally, I was recently exposed to a pair of rather large loudspeakers packaged in a novel way that could conceivably overcome spousal objection, even though at first glance the speakers actually appear to be larger than they are. The system, not yet sold anywhere, is called the **Applied Acoustics Model 10A** and is the brainchild of Jim Suhre, a Raytheon rocket scientist (really!), and Vic Kalilec, an electronics engineer, both from Tennessee. What they showed me was a monumental floor-to-ceiling wall system, the most salient feature of which is its gigantic, seamless, curved tambour door. Open the door and all your electronic equipment is in there, including your large-screen TV. Close the door and the whole shebang looks structural, not like audio equipment. Jim Suhre claims that the curved tambour acts as an ultrasophisticated dispersion device. I have my reservations about that but can report that the sound had exceptionally even spectral balance, from the lowest to the highest frequencies. The speakers, when their floor-to-ceiling grille is taken away, are revealed to be only chest high; I'd say they have the overall impact of B&W Matrix 803's or something along those lines. The drivers and network appear to be of the highest quality. The price was still up in the air when I looked; those who go for this sort of thing will no doubt be able to afford it.)

ACI "Spirit"

Audio Concepts, Inc., 901 South 4th Street, La Crosse, WI 54601. "Spirit" floor-standing 2-way loudspeaker system, $499.00 the pair (direct from ACI). Tested samples on loan from manufacturer.

"In this world nothing can be said to be certain, except death and taxes," Ben Franklin wrote. That may have been true in 1789, but today I would add: "and good value from ACI." The combination of direct marketing, conscientious design, and just plain common sense have made Mike Dzurko's trademark synonymous with "more speaker for your dollar." That's certainly true of a pair of Spirits, which satisfy all the basic audiophile demands, except for the deepest bass, at the totally unexpected price of $499.

The speaker is a 32" high box with a footprint of less than a square foot, housing an 8" woofer with polypropylene cone and a 1" aluminum-dome tweeter. The woofer is aperiodically loaded with four small holes in the back of the cabinet near the floor; the tweeter has a plastic dispersion plug and is surrounded with felt. The left and right speakers are mirror-imaged. The oak veneer of my samples was of good quality.

The impedance curve of the Spirit shows the box to be tuned to 50 Hz and the minimum impedance of the system to be 6½ ohms (8 ohms nominal). Total impedance variation is between that minimum and 33 ohms in magnitude and ±45° in phase. Any decent amplifier should be able to drive such a load.

The quasi-anechoic (MLS) frequency response was interesting in that both the woofer and tweeter were quite flat, within ±2 dB or so, but the transition between them in the 2 to 3.5 kHz range was not smooth, showing various irregularities of the order of 7 dB, depending on the angle of measurement. Since all these irregularities were basically minus (i.e., not peaks) and in the most sensitive range of the ear, they may have acted as inadvertent zippiness suppressors. At this price, you can't expect the crossover network to be too sophisticated; it appears to be second-order, with out-of-phase wiring of the drivers. The tweeter has a double resonance at 17 kHz and 24 kHz; otherwise it's quite smooth and remarkably clean in response to tone bursts; there is no ringing at any frequency. The same is true of the woofer cone.

The nearfield response of the woofer shows the f_3 (-3 dB point) to be 50 Hz, with only a 12 dB per octave rolloff below that frequency. That means useful response down to 35 Hz or so. Can't ask for much more than that.

The sound of the Spirit is essentially neutral. That's a simple statement but not a simple achievement. More than a few extremely costly speakers don't sound neutral. Transparency is good but not superb (what did you expect?). Dynamic range is very good. I didn't measure the distortion because it's obvious from the drivers that it can't be either very low or very high, and the process is time-consuming; the sound, let me assure you, is quite clean at high levels. All in all, this is a highly acceptable, far from puny-sounding, musically pleasing little loudspeaker. I was demoing a very high-end speaker to a friend, and then switched to the Spirits. The difference was obvious but not very dramatic. Hey, for $499?

Bag End ELF Systems S10E-C and S18E-C
(continued from Issue No. 21)

Modular Sound Systems, Inc., P.O. Box 488, Barrington, IL 60011. Voice: (708) 382-4550. Fax: (708) 382-4551. ELF-1 two-channel dual integrator electronics, $2460.00. S10E-C black-carpet enclosure with single 10" woofer, $234.00 each. S18E-C black-carpet enclosure with single 18" woofer, $658.00 each. Tested samples on loan from manufacturer.

I see this equipment in a slightly different perspective now that I have measured it and evaluated it at considerably greater length. I don't take back my original statement that "I have never heard bass like this in my listening room," but the reason for that was not the ELF technology. It was the combination of the air-moving capability of two 18" drivers, flat response all the way down into the 20-to-10-Hz octave, excellent damping, and little or no dynamic compression—a combination I hadn't previously experienced, as a total package, in my listening setup.

I now believe that good conventional technology could achieve the same results—even if it rarely does, for various reasons. (See also David Rich's sidebar on the subject and the Velodyne Servo F-1500R review below.) Since an ELF-1 unit plus two S18E sub woofers plus two high-powered amplifier channels could run into $5000 or more, the question is whether or not the ELF approach yields any substantive benefits in a domestic sound system (as distinct from professional applications—see below) that are not obtainable by simpler means for less money. My present feeling is that, from the stay-at-home audiophile's point of view, the ELF system is "a solution in search of a problem." That doesn't make it sound less good than I said, but the I've-got-to-have-it factor is gone because what makes it sound good isn't its uniqueness.

My measurements showed the S10E and S18E to be almost identical in small-signal frequency response when driven via the ELF-1 (with all switches down, the most wide-open setting). The deviation from absolute flatness is ±0.5 dB down to 20 Hz, dropping to -4 dB at 10 Hz. The Bag End literature shows curves for the S18E that indicate -2 dB response at 10 Hz, which is more consistent with an f_3 (-3 dB frequency) of 8 Hz, the claimed system cutoff. I see no reason to make a federal case out of the discrepancy; the measured f_3 of 12 Hz is good enough for me. Of course, as the signal is increased, the air-moving and power-handling capability of the S18E quickly passes that of the S10E; on the other hand, multiple S10E's in a cluster could keep up with the S18E in all respects.

The f_3 also moves up, inevitably, as the level is increased; either the driver or the amplifier (remember, it's boosted 12 dB per octave), or both, will reach a limit in linear output capability. The ELF concealment circuit deals with this very neatly (again, see sidebar); it is probably the cleverest and most original element of the total system. You can crank the volume to any level you wish; at some point when, say, the bass drum is thwacked, the concealment threshold lights come on, but you hear no distortion and are unaware of compression; it's smooth as silk. You could argue, of course, that this is needed only because the system has the inherent weakness of being based on tremendous electronic boost—one complicated mechanism to correct the side effects of another complicated mechanism.

I measured the distortion of the Bag End subwoofers with the ELF concealment threshold set to leave the circuit inactive at the levels tested (all switches down). That way I was measuring the true electroacoustic accuracy of the equalized transducers. Needless to say, the amplifier was at all times operating well below clipping.

On the whole, the distortion figures were not impressive. For example, at 30 Hz, as I gradually increased the 1-meter SPL from 80 dB to 95 dB, the distortion of the S18E as measured very close to the cone went from 2% up to 4%. At 40 Hz, the distortion over the same SPL range varied between 1.1% and 1.7%. At 20 Hz, I measured 2.6% (80 dB) to 10% (95 dB). Compare that with the Velodyne F-1500's distortion figures (see below) and you'll begin to understand the design priorities of each. The S10E has a very similar distortion profile down to 30 Hz but gets worse at 20 Hz, as you'd expect. Overall, even the bargain-priced Hsu Research HRSW10 subwoofer (Issue No. 19, pp. 21-22) beats the Bag Ends on

What the Bag End ELF System Does and Doesn't

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

OK, here we are stuck in a sidebar because of all the equations I want to use. But do not worry; it really is not so complicated. The first thing we need is the transfer function of a sealed-box-loaded speaker:

$$\frac{s^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}$$

The amplitude and phase of a loudspeaker's output are related to its input signal through the system function that governs its operation. For a sealed-box system, this turns out to be a second-order differential equation. The expression above uses the Laplace transform, which relates time functions to frequency-dependent functions. What it is going to allow us to do is to manipulate the differential equations using just algebra. (Yes, I realize that those who are sophisticated in mathematics have just gone into clipping over this oversimplified statement.) The expression above is a second-order highpass filter. Above the frequency w_0 the output signal level is the same as the input. Below w_0 the signal output declines at 12 dB per octave. This w_0 is the -3 dB cutoff frequency in radians per second (Hz x 2π) that you already know about. The Q in the equation is the Q you have seen mentioned in this magazine many times. The value of Q determines the shape of the response near w_0. If Q is above 0.707, the response will have a peak before rolloff. Now let us assume that you want to put in series with the loudspeaker an electronic equalizer that would change the value of w_0 to a lower value w_1. Physically the equalizer boosts the input signal to the speaker between the range from w_0 down to w_1 The transfer function of an equalizer that will do this is given below.

$$\underbrace{\frac{s^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}}_{\text{Original loudspeaker response}} \times \underbrace{\frac{s^2 + \frac{\omega_0}{0.707}s + \omega_0^2}{s^2 + \frac{\omega_1}{0.707}s + \omega_1^2}}_{\text{Electronic equalizer}}$$

$$= \underbrace{\frac{s^2}{s^2 + \frac{\omega_1}{0.707}s + \omega_1^2}}_{\text{New loudspeaker response if } Q = 0.707}$$

An electronic equalizer that generated the required response was a product from Allison Acoustics, called the Electronic Subwoofer. This device came out in 1978. It is no longer available, perhaps because it only worked with sealed loudspeakers that had a Q of 0.707 and it required the user to know the value of w_0. The other problem with the Electronic Subwoofer was that it could boost power into the speaker by more than 10 times, depending on the ratio of w_0 to w_1 This required a powerful amplifier and a loudspeaker with low distortion characteristics below its cutoff frequency w_0

The ELF Concealment system used by Bag End neatly solves the last problem by making the value of w_0 dependent on the power level. The value of w_1 is increased as the power level of the signal becomes higher in the range below w_0. This prevents the amplifier from clipping and prevents the subwoofer from distorting or bottoming out. The method by which w_1 is moved is proprietary, but I would expect it is similar to the methods used by Dolby Laboratories in the Dolby noise-reduction systems. Now w_1 can be set to have a bottom limit at a very low frequency. In the case of the Bag End ELF systems this is around 10 Hz. The result is very extended low-frequency response and also very low phase shift at the higher frequencies where the subwoofer will be crossed over. This makes it easier to cross the speaker over without amplitude variations in the crossover region due to subwoofer phase shift. Because w_1 can be set very low, the Bag End ELF system adds an additional second-order lowpass filter fixed at 10 Hz in order to prevent subsonic overload. (The Allison Electronic Subwoofer also had a subsonic filter.)

The Bag End ELF system works a little differently than the basic configuration I started this explanation with. The modification involves setting the subwoofer's w_0 to a very high value of, say, 440 rad/sec (70 Hz). This modification permits reduced size or improved efficiency (which limits cone movement) or a negotiable combination of both. Since the equalizer is going to introduce significant boost, this is clearly an important feature, but the advantage is somewhat reduced by the fact that the maximum boost at low frequencies (w_0/w_1) is now higher, since w_0 is larger.

The equation to equalize the Bag End woofer is given below.

$$\underbrace{\frac{s^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}}_{\text{Original loudspeaker response}} \times \underbrace{\frac{\omega_0^2}{s^2 + \frac{\omega_1}{0.5}s + \omega_1^2}}_{\text{Electronic equalizer}}$$

$$= \underbrace{\frac{s^2}{s^2 + \frac{\omega_1}{0.5}s + \omega_1^2}}_{\text{New loudspeaker response}} \times \underbrace{\frac{\omega_0^2}{s^2 + \frac{\omega_0}{Q}s + \omega_0^2}}_{\text{Lowpass filter to be used for the crossover}}$$

As can be seen, in the Bag End ELF approach an additional term appears on the right-hand side of this equation. This is a second-order lowpass filter with a cutoff frequency of w_0 (C. 70 Hz for the Bag End subwoofer) and a Q of 0.707 (that of the Bag End subwoofer). The equalizer has a $Q = 0.5$ because it is formed by cascading two first-order equalizers.

The ELF electronics contain a matching second-order highpass filter to be cascaded with the main loudspeaker. The cutoff frequency of this filter can be varied from 50 Hz to 205 Hz in 5 Hz increments. If the main speaker had a cutoff well below 70 Hz—but in that case why do you need a subwoofer?—then the highpass filter would be set to 70 Hz for flat frequency response. But we have a problem, since this requires that the w_0 of the subwoofer, which is based on mechanical parameters, should match the electrical parameters of the highpass filter. Any mismatches and frequency response errors will occur around the crossover frequencies. The same problem exists with the Allison Electronic Subwoofer, but here the frequency aberrations will occur at the cutoff frequency of the main loudspeaker, typically around 40 Hz. The ear will be less sensitive to this than in the region of 70 Hz.

But a bigger problem is that the highpass crossover of 70 Hz is not going to be correct for many kinds of main speakers, especially not for small monidistortion. To say something nicer, the damping of these sub woofers is superb; low-frequency tone bursts in the 30 to 70 Hz range excite virtually no spurious output, regardless of the number of cycles in the burst. (That's what the untutored would call "fast.")

As for impedance, both the S18E and S10E are essentially 6-ohm units, but the fundamental resonance, which determines the ELF lowpass "corner" frequency, is slightly different for each: 62 Hz for the S18E, 67 Hz for the S10E, at least in my samples. Sample-to-sample variations can be assumed; in a general discussion of the ELF system, 70 Hz would be a safe round number to use.

Since the ELF-1 unit acts as a front end just before the power amplifiers, I measured it as if it were a line-level preamplifier. At both ELF Out and HF Out in each channel, the THD + N versus level curves appeared to be noise-dominated, as in any decent preamp. The minima at both outputs and at most frequencies were in the -81 to -87 dB range, with soft clipping before the 10 V absolute maximum output. That's par for the course. The 20 kHz distortion profile at HF Out, however, wasn't very good. Even with the most favorable settings, the 20 kHz distortion was at its minimum of -74 dB at only 650 mV out, then climbed steeply to -60 dB at 4 V out and worse beyond that. That's a classic case of dynamic distortion. Since no circuit schematic was available, only a block diagram, I didn't involve David Rich in diagnosing the possible causes of this, but it doesn't really matter. We aren't dealing with audible levels of distortion here but only engineering refinements. Still, why not do it 100% right? Channel separation at HF Out was 45 dB at 20 kHz, 66 dB at 1 kHz, and 97 dB at 20 Hz. Again, that's OK, but just. The ELF-1 doesn't scale new heights as an analog front end.

My present conclusion about the Bag End ELF approach, now that all the tests and analytical comparisons are behind us, is that it probably still works best in professional sound, where the whole thing started in the first place. A rock group, for example, could set up large clusters of the highly portable and cost-effective S10E and possibly get away with using only a single expensive ELF-1 to drive all the amplifiers. That makes sense. For residential audio this is not the system of choice as I now see it. Of course, if you've already bought it, you've got one hell of a subwoofer setup, capable of stupendous bass, but you could have obtained equal or possibly even superior results by simpler and less costly means.

Velodyne DF-661

(continued from Issue No. 21)

Velodyne Acoustics, Inc., 1070 Commercial Street, Suite 101, San Jose, CA 95112. Voice: (408) 436-7270. Fax: (408) 436-7276. DF-661 "Distortion-Free Full-Range Loudspeaker," $1695.00 the pair in black vinyl, $2245.00 the pair in rosewood. Tested samples on loan from manufacturer.

After several months of exposure to this radically new and different speaker system, and after having completed the usual laboratory measurements (and more), I must reiterate my initial impression that we are dealing with a very important, groundbreaking design here. It has flaws, but its virtues are infinitely greater. There may be a problem, however, with that evaluation. It is possible, though by no means certain, that our review pair was measurably and audibly superior to the samples tested by other reviewers (John Atkinson for *Stereophile,* Julian Hirsch for *Stereo Review,* David Moran for the Boston Audio Society, et al.). David Hall, president of Velodyne and designer of the DF-661, told me that there had been some early QC problems with the aluminum-cone drivers; conceivably we lucked out and got a perfect pair. Now, an engineering design is as good as the most perfect available sample, but a commercial product is only as good as the least perfect sample that leaves the factory, and that's something I can't possibly pass judgment on.

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tors, since they may already be rolling off by then. And, most subwoofers should be crossed over with a slope of at least 18 dB or 24 dB per octave. Higher-order crossover slopes limit the range over which the speakers interfere with each other and reduce the amount of low-frequency energy sent to the main speaker. Given the unchangeable 12 dB per octave slopes of the ELF crossover, the ideal match for the subwoofer would be a main speaker with flat response down to at least 35 Hz, so that in the 70 Hz crossover region the highpass filter alone would control the main speaker's low-end rolloff.

Both the Bag End ELF woofer and the Allison Electronic Subwoofer are examples of open-loop correction systems. A closed-loop correction system using feedback is clearly a more sophisticated way to extend low-frequency response and reduce distortion. The problem with a closed-loop system is the transducer (motion sensor) required to convert the acoustic signal back to electrical energy. Any errors in the transducer's transfer response will be reflected in the speaker's closed-loop response.

It is unfortunate that the good basic design work by the engineers of the Bag End ELF system is tarnished by the literature supplied by the company. The literature makes claims for the system that are not true, and more often than not misstates or oversimplifies how the system works. A good example of the oversimplification is the use of the concept of an *integrator* to explain how the equalizer in the subwoofer path works. That's not how it works, as explained above. An example of complete inaccuracy is the claim that the signal delay in the Bag End ELF system is short and fixed because integrators have fixed delay. As can be seen from the above, the delay is not any different from that of any other loudspeaker with the same low cutoff frequency.

DF-661—from top to bottom: 6" midrange driver, 1" dome tweeter, 6" woofer—requires aiming the measuring microphone at the tweeter for correctly summed on-axis and off-axis measurements. The quasi-anechoic MLS technique yielded much flatter response on axis, both at 1 meter and 2 meters, than reported by others. The tweeter (made by LPG but labeled by Velodyne) appeared to be almost amplifier-flat above 7 kHz, right up to the ultrasonic resonance. Below 4 kHz, the response is again extremely flat, within ±2 dB down to 300 Hz. Only between 4 kHz and 7 kHz is there a larger deviation, namely a 4 to 5 dB dip centering approximately on the 5 kHz crossover frequency. This dip becomes a deep and wide notch, of the order of 6 to 8 dB, in the off-axis area of radiation. A nonoptimal crossover network is probably the culprit, possibly aggravated (and certainly not helped) by diffraction from the sharp edges of the cabinet. The crossover schematic shows that the tweeter is brought in with a very steep (24 dB per octave) highpass filter, ostensibly to restrict the moving system to its linear range, whereas the midrange driver is rolled off on top with a 12 dB per octave filter. There may be a mismatch between the two filters. Another possibility is that the midrange driver is simply too large to have good response up to 5 kHz over a reasonable solid angle. Even so, the DF-661 still has a better frequency-response profile than the majority of speakers, at least on the basis of our samples. Except...

Except, yes, for the bass. There isn't much of it. The vented-box system appears to be tuned to something between 60 and 62 Hz, but it is a very peculiar kind of tuning, further confused by a 250 µF capacitor in series with the woofer. The purpose of the capacitor is to restrict the excursion of the driver and keep the distortion very low, but it also makes the design parameters of the bass system hard to determine with standard measurements. In any event, this is at best a weakish 60 Hz box, meaning that a subwoofer is needed for full-range performance, and that makes the funny bass more or less moot, at least in my book.

The impedance curve of the DF-661 is quite reasonable above 100 Hz, fluctuating between 3.2 and 8.5 ohms in magnitude, and remaining within ±30° in phase (the latter only above 200 Hz). At the lower frequencies that huge capacitor takes over; the magnitude goes way up and the phase goes highly negative—what else did you expect?

So why am I so enthusiastic about the DF-661? Because, to my ears, it sounds more transparent, more accurate in rendering instrumental and vocal timbre, more lifelike in overall tonality than any other speaker I was able to compare it with, either side by side or from memory. The reason for that is almost certainly the ultralow distortion, which was David Hall's main design goal. In fact, he explained to me that he had paid only routine attention to other parameters and focused almost entirely on distortion "because I wanted to prove a point." I find his point proven because the DF-661 is merely a decent but not great speaker in all other respects and yet it sets a new standard in transparency. It must be the low distortion; there's nothing else to account for it.

How low? I measured the woofer and the midrange driver separately, each driven with a continuous signal to a midband level of 95 to 96 dB at an axial distance of 1 meter. (That's loud!) The distortion reading was in the nearfield, right off the diaphragm. The midrange driver did not exceed 0.2% THD + N at any frequency between 800 Hz and 4 kHz, remaining well below 0.1% over most of that range and dipping as low as 0.04%. That's more like an amplifier than an electroacoustic transducer. The woofer averaged 0.1% in the 125 to 800 Hz range, dipping as low as 0.03%, but climbing below 125 Hz to a peak of 1.3% at 63 Hz. Pretty damn impressive—unless you think, like some nitwits, that distortion matters only in the electronic signal path but not the speaker. Well, my ears tell me that it matters a great deal.

Please note, however, that with my priorities I represent one kind of listener; as I wrote in the last issue, "I am basically a tonality/balance/transparency man." You may be an imaging man, or a front-to-back depth man (or woman—sorry), in which case *the* DF-661 may not be your cup of tea. The soundstage it presents is a little flat in comparison with some speakers (e.g., the Win SM-8, reviewed below). The reason may again be the crossover problem or diffraction or both; indeed, the deep crossover notch off axis may be part of the syndrome. As I said, the design is not without flaws. The network, by the way, drives the midrange out of phase with woofer and tweeter; the second-order (12 dB per octave) slopes of the midrange bandpass filter appear to be the reason why.

The drivers themselves, in addition to producing extremely low harmonic distortion, are also very clean in terms of freedom from ringing. I made more tone-burst measurements, at more frequencies, than I ordinarily do and found only negligible storage. Whatever weaknesses the design may have are all related to frequency response. (Of course, as I said before, I can only talk about the particular pair I tested.)

Bottom line: I played one of my favorite Julianne Baird tracks (on Dorian) through the DF-661, then switched to a speaker I had previously thought of as quite wonderful. It sounded disturbingly veiled by comparison. (I won't identify the speaker because the same thing would have happened with any number of others.) The more I played DF-661, the more addicted I became to its tonality/transparency characteristics and began to find other, in some ways better, speakers unsatisfactory. What I currently listen to most often—just for enjoyment that is, when I'm not testing—is the Velodyne DF-661, extended on the bottom with the Velodyne Servo F-1500R subwoofer. (A separate review of the latter follows.)

Final thoughts: I have been told that the unfavorable review in the June 1994 issue of *Stereophile* pretty much shut down the sales of the DF-661. Now, I consider *Stereophile* quite capable of "cooking" their measurements and reporting worse sound than they actually heard, because ultrahigh-end performance out of a lowly upper-medium-priced speaker simply cannot be allowed —it's against their religion. (Remember the number they did on the Carver "Amazing"?) Such ugly thoughts must be quickly reconsidered, however, in the light of all the other not-so-enthusiastic reviews. It's hard to believe they're all just politicking. My recently calibrated Brüel & Kjær 4133 microphone and Audio Precision "System One Dual Domain" are too dumb to lie, and the evidence of my ears was corroborated by several staff members, so I must stick to my guns; even so, as I pointed out, our samples may conceivably have been better than what everybody else got. More than that I cannot say.

Velodyne Servo F-1500R

Velodyne Acoustics, Inc., 1070 Commercial Street, Suite 101, San Jose, CA 95112. Voice: (408) 436-7270. Fax: (408) 436-7276. Servo F-1500R servo-controlled powered subwoofer with remote control, $1595.00 each (3190.00 the pair). Tested samples on loan from manufacturer.

In Issue No. 16 I made it quite clear that I have the highest respect for Velodyne's motional-feedback approach to bass transducer design. Now that I have tested their latest-and-greatest model—not just a single unit but a pair—I feel even more strongly that they're doing it right. Their distortion figures are so much lower than anyone else's that I find myself unable to favor designs with higher distortion regardless of other possible advantages. For example, in the case of the F-1500R the small-signal bass-cutoff frequency (f_3) is 16 Hz as against the Bag End S18E's 12 Hz. That's not enough of a difference in low-frequency extension, at least in my opinion, to opt for the far costlier Bag End ELF solution with its considerably higher distortion. Remember, distortion is the difference between the input and the output, i.e., the index of accuracy—and accuracy is what high fidelity is all about.

I bring up the Bag End ELF system specifically because it was my top choice (see the review above) before the Velodynes arrived. A pair of Bag End S18E's will move slightly more air than a pair of F-1500R's, which are "only" 15-inch woofers, but the subjectively perceived heft, ease, and lifelikeness on the bottom end are very similar in a comparison of the two designs, and the Velodyne then moves ahead of the Bag End on other counts. The far lower distortion is only one of these; another is the convenience of the built-in 250-watt power amplifier; still another the variable lowpass filter, adjustable from a nominal 40 to 100 Hz, allowing the flexibility to match various highpass contours for the satellite speaker. With the ELF system you are married to a single crossover frequency, namely the fundamental resonance of the woofer (typically 60 to 70 Hz). All in all, I now consider the Velodyne to be a more desirable subwoofer for domestic music systems—but that doesn't make the ELF system less impressive than I originally reported.

Measuring the distortion of the F-1500R requires awareness of its compressor/limiter circuit, which holds the output level below a certain distortion limit, regardless of the input level. When the limit is reached, increased input will not result in increased output. That in itself is a good thing—what's the point of higher output if it's distorted?—but it restricted my full-bandwidth measurements to a maximum SPL of 95 dB at 1 meter (normalized to 40 Hz). At that level—which is damn loud!—there appears to be no limiting at any frequency. THD + N at 95 dB hovered between 0.12% and 0.2% from 100 Hz down to 28 Hz; at 20 Hz it rose to 0.6%; at 16 Hz it was 0.8%. At 90 dB the 20 Hz distortion was only 0.24% but above 30 Hz the THD + N matched quite closely the 95 dB profile. Going lower in SPL actually increased the measured percentage of distortion slightly, probably because of the irreducible amount of ambient noise picked up by the microphone. Overall, this is sensationally low distortion, unequaled by any woofer design known to me. Just for the record, Velodyne provides the following distortion spec, which I didn't verify but find compatible with my measurements: less than 0.7% at 25 Hz at 104 dB SPL. Would it be possible to obtain comparable distortion figures without motional feedback? David Hall told me yes, with the same transducer design techniques he used in the DF-661; so far, however, he hasn't applied those techniques to a big woofer, and lately he has expressed doubts about the feasibility of it. As for flatness of frequency response, I measured ±0.25 dB over the unit's intended range—but you would expect that of any decent design.

It should be noted that the F-1500R is a forward-firing 15" subwoofer, unlike the ULD-15 Series II reviewed in Issue No. 16, which has a similar 15" driver firing downward. A forward-firing cone has only mass; a downward-firing cone has both mass and weight. The weight (i.e., the pull of gravity) can cause stress on the suspension and eventually affect proper centering, although techniques exist to counteract that problem. Also, the straight-ahead wave-launch is more predictable and controllable than floor loading. All in all, I'm more comfortable with the forward-firing configuration, and so are most speaker designers. The R suffix stands for remote control, the latest wrinkle added by Velodyne. The slim, handheld remote unit controls power on/off, volume down, volume up, volume restored to manual setting, and mute. To me this is a very minor convenience, but some will consider it to be totally cool, and they are entitled to their gizmos, playthings, and sitting habits.

Does the F-1500R have a weakness? A single-cycle 25 Hz tone burst into the amplifier input results in

1½ cycles of acoustical output, indicating a very slight problem with damping at that worst-case frequency. At higher frequencies I found no such (mini)hangover. No big deal, in any event.

In my old ULD-15 review referred to above I expressed my wish for a pair of subwoofers instead of a single matrixed (L + R) unit and gave reasons. Our columnist Tom Nousaine, for one, disputes those reasons and has some good arguments to the contrary, as well as experimental data. Others, including David Hall, side with me. This is a legitimate controversy (unlike so many stupid tweako hassles in the high-end press) and merits fuller treatment, perhaps even a complete article. Meanwhile I'm as happy as the proverbial pig in excreta with my, count them, *two* F-1500R's. Matrix, schmatrix, I refuse to part with either one of them.

Win SM-8

Win Research Group, Inc., 7320 Hollister Avenue, Goleta, CA 93117. Voice: (805) 968-5213. Fax: (805) 685-2781. SM-8 Studio Monitor, $5000.00 the pair, including stands. Tested samples on loan from manufacturer.

This is a derivative of the marvelous Win SM-10 reviewed in Issue No. 17—not better, not less good, but an alternative design with somewhat different tradeoffs—and the reader is referred to that review for (1) my comments on Dr. Sao Zaw Win as an audio designer, (2) the mathematical credentials behind the Win speaker-design efforts, and (3) a complete discussion of the black lacquer cabinet, which is identical in both models.

The immediate *raison d'être* of the SM-8 is that the SM-10 is no longer being made, at least until a legal hassle between Win Research and Matsushita (Panasonic) has been resolved. (The SM-10's diaphragms, magnets, and voice coils were Win designs; the frame, however, was based on Matsushita tooling, which Win had purchased exclusive rights to—but Matsushita apparently thought otherwise. The controversy is too convoluted for further comment here.) The SM-8 adheres to the same rigorous design philosophy as the SM-10 but instead of the latter's coaxial configuration it uses a conventional woofer/tweeter layout. As a result, coherence in the nearfield is not as perfect, but there are also advantages, as we shall see.

The new 8" woofer of the SM-8, covering the range up to 2 kHz, is considered by Sao Win himself to represent his best work to date. It has a flat diaphragm made of molded balsa-wood pulp, a completely new idea. Flexing is the major problem with all speaker diaphragms above a certain size, and this solution is claimed to be a breakthrough in that regard. Other diaphragm materials represent various kinds of compromise between stiffness, mass, and internal damping; balsa combines a very high stiffness-to-mass ratio with good self-damping characteristics and is claimed by Sao Win to be a no-compromise substance for the purpose. The balsa is pulped, and the slurry is poured into an intricate gridlike mold, designed with computer simulation techniques and finite element analysis to obtain the maximum possible structural strength in the resulting flat membrane. It isn't your usual speaker cone, that's for sure.

The tweeter is a highly customized inverted-dome Accuton, using the magnet structure of their *3½* driver but the diaphragm of their 1" unit. Special doping of the ceramic dome in accordance with Sao Win's preferences is part of the OEM custom specifications. Again, it's not just a tweeter out of the catalog.

The crossover network is in a separate, very handsome case with a transparent cover. The fully visible circuit board is a work of art. The network is a modified 4th-order Linkwitz-Riley, computer-optimized to include the acoustical rolloff characteristics of the drivers. It's the right choice in my opinion, especially since the 2 kHz crossover frequency is quite close to the woofer's upper and the tweeter's lower limit for best performance. The design of a two-way system almost inevitably stretches those two limits in search of the best possible handshake between the two drivers. (That's why there are three-way systems.) Steep crossover slopes are a big help under such circumstances.

The integral speaker stand of the SM-8 is a definite improvement over the SM-10 stand. Made of a heavy alloy chosen for exceptional acoustical deadness, this black tubular stand is very easy to mount and tilts the speaker backward at a slight angle. The benefits of the latter feature are debatable, but of course it does offer a wee bit of delay compensation and looks high-tech.

As I said, the vented cabinet is identical to that of the SM-10; the costly finite element analysis that had been originally invested in that structure had to be saved, so the woofer parameters were adjusted to those of the enclosure rather than vice versa. My raving endorsement of this gorgeous box can be found on page 13 of Issue No. 17.

Since there are separate input terminals for the two drivers to accommodate the external crossover network, I was able to measure the "naked" woofer and tweeter. The woofer appears to have flat response (within approximately ±2 dB) and no anomalies over its assigned range, except for a slight rise just before it starts rolling off very steeply a 2 kHz. That rolloff point is an Achilles' heel; a 2 kHz tone burst excites significant ringing, not evident at any other frequency. The crossover notch completely hides this weakness when the network is connected. On the bottom end, the vented box is tuned to 41 Hz and the maximum output of the vent is at 53 to 54 Hz; that's deeper bass than is obtainable with the SM-10 and is about all that can be squeezed out of any 8" driver in an enclosure of 0.75 cubic foot internal volume without seriously compromising efficiency. With just a little bit of generosity this can be called a 40-Hz box. (You can't deny that computer optimization has greatly helped the design of vented systems.)

The tweeter, without the network, exhibits a somewhat puzzling response profile. On axis, up to 7 kHz, it's flat within ±1 dB. Between 8.5 and 20 kHz it's even flatter, ±0.5 dB or at worst ±0.75 dB. Between 7 and 8.5 kHz, however, there's a 4 dB step downward. Very strange. Off axis, still without the network, the step is "homogenized" into a smooth 3-dB-per-octave rolloff above 5 kHz (as measured at 30°). Also strange, since the 20 kHz response is then barely down from the on-axis level. But wait. Inserting the crossover network changes the picture.

The network has a tweeter level control, which I left in the 0 (all the way up) position for my measurements. The on-axis response of the complete system is virtually identical whether the microphone is aimed at the tweeter or at the midpoint between the woofer and tweeter centers. That response is a smooth downward slope for a total drop of 6 dB from 2 to 20 kHz. Going 30° off axis barely changes this measurement, at least up to 12 kHz. Between 12 and 20 kHz there is an additional drop of 8 dB, but that has little effect on the subjectively perceived high-frequency energy in the off-axis area. So what we have here, instead of the usual flat response on axis and significantly rolled-off response off axis, is an almost invariant, gently sloping response over a large solid angle. For all I know, the true power response of the SM-8 is flatter than that of a more typical "flat" speaker. With my 1-meter MLS measurements, all I can be sure of is the anechoic response at a limited number of microphone positions; I don't have an averaged power response. Certainly, the SM-8 doesn't sound attenuated on top; if anything, it may sound a little bright on some program material with the tweeter level set to 0, in which case the -1 setting should be tried.

Tone bursts reveal no anomalies in the drivers other than the woofer's minor problem at 2 kHz noted above. The bottom end of the system could be slightly better damped, but then the bass would be leaner and the quasi-full-range impact of the speaker diminished—it's a tradeoff. Distortion is extremely low in the tweeter; in the crucial 2 to 5 kHz range, at a 1-meter SPL of 95 dB, THD + N stays between 0.1 and 0.2%. That's almost as good—at some frequencies just as good— as the distortion performance of the Velodyne 6" midrange driver over the same range, proving that the 1" tweeter is not being "stretched" excessively at its lower frequencies. The woofer is also impresively low in distortion but not quite as low as the Velodyne 6" low-frequency driver. In the range from 50 Hz to 2 kHz, the Win woofer fluctuates between 0.25% and 1.2% THD + N at a 1-meter SPL of 95 dB. Ignoring the widest swings, the distortion at that hefty level can be said to stay mostly within 0.45% to 0.75%. That still makes the SM-8 the silver medalist in distortion competition, I think. The impedance curve of the system is classic and unproblematic; nominal impedance is 6.5 ohms.

That brings me to the evaluation of the sound (since you can't stand the suspense any longer). I didn't fall in love with the SM-8 as I did with the SM-10, even though the SM-8 can be driven harder, is cleaner at high levels, has deeper bass, and images just precisely. The SM-10 was no longer available, so there was no possibility of a side-by side comparison, but somehow the exquisite beauty of the string sound, the silky, delicate, finely etched quality I remembered, seemed to be absent, at least to some degree. That's *my* subjective impression; not everyone will agree with me. According to Sao Win, the mica-and-alumina tweeter of the SM-10 is even better than the Accuton custom unit but will not extend down below 2 kHz as is required in the SM-8 design. He is working on a solution and the change will be made—if and when. The SM-8, having been designed to the same listening standards as the SM-10, sounds very similar of course, and some will probably prefer it for the aforesaid advantages. As for the coaxial versus up-and-down configuration, the latter permits larger magnet structures and generally leaves more elbowroom for massaging the driver designs, so the SM-8 trades away perfect coherence for increased large-signal capability. I don't hear anything negative in the sound of the SM-8 that I could specifically attribute to the absence of coaxial geometry.

If you have read the review of the Velodyne DF-661 above, you can imagine that I wanted a shootout between it and the SM-8. As it turned out, the two were almost perfectly matched in their farfield volume level without any attenuators. Two associates and I spent at least two hours each switching back and forth between the two speakers, playing all kinds of program material. There was enough of an instantly audible difference to obviate blind testing but not enough to result in a quick and definite preference. The Win has better bass and images far better. The Velodyne excels the Win in transparency and naturalness of timbre. That was the consensus, but we all agreed that the two are close in quality. I continued the comparison for several days without significant new insights to add. Visually there's no comparison; the Velodyne looks just fine but the Win is a stunning showpiece. On the other hand, the Velodyne costs less than half as much, and that makes it the easy winner on a per-dollar basis. It's a nice dilemma, when you think about it.

Note: *None of the above applies to a few pairs of early SM-8's with very low serial numbers, which had a different crossover network and different tweeter.* •

Good Things Are Still Happening in Analog Electronics

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

Preamplifiers and power amplifiers are getting to be so uniformly good that reviewers will soon have relatively little to review.

I laugh my head off these days reading the reviews of preamps and power amps in the tweako magazines. Soundstaging, front-to-back depth, midrange transparency, brightness, dullness—all the pluses and minuses of microphones, recording methods, speaker systems, and listening rooms are attributed to these poor electronic signal paths, which have demonstrably no sound of their own unless very poorly designed or downright defective or overdriven. They still need competent reviewing, of course, because poor design is always an outside possibility, distortion usually an index of engineering sophistication, adequate power into difficult loads often a need, build quality always a measure of value per dollar, ergonomics invariably a livableness factor, etc., etc.—I have been over this same ground a number of times.

The large accumulation of well-documented double-blind listening tests offers overwhelming proof that fanatically intense "golden-ear" comparisons of this type of equipment are a total waste of time. The may sell magazines to the wishful-thinking nonprofessional but yield no genuine consumer benefits. We still listen very carefully to each preamp and power amp we review because it's a simple and direct way to spot anything unexpected, but please don't expect us to dwell lovingly on the upper-midrangeliquidity and other bathroom-literature themes. The Audio Precision System One has the most golden ear of any reviewer.

—Peter Aczel

Line-Level Preamplifier

Aragon 18k

(Reviewed by Peter Aczel)

Mondial Designs Limited, 20 Livingstone Avenue, Dobbs Ferry, NY 10522. Voice: (914) 693-8008. Fax: (914) 693-7199. Aragon 18k line-level preamplifier, $995.00. Tested sample on loan from manufacturer.

I must refer the reader to David Rich's review of the Acurus L10 in Issue No. 18 (page 21). The Aragon 18k is basically an improved version of that $595 "best buy"—for a lot more money ($1.67 on the dollar, to be precise). For even more money, you can specify the Penny & Giles volume control, an extremely high-quality potentiometer.

What are the improvements? Current sources have replaced the resistors to bias the differential pairs, for even lower distortion. MOSFETs running higher bias currents have replaced the JFETs, allowing full drive into loads of lower impedance. The outputs are now direct-coupled, with a trim pot to cancel dc offset. Power rails are up by 2 V to ±20 V. The beefed-up power supply is now an outboard unit in a vaguely pyramidal (New Age?) metal box. The input and output jacks are no longer mounted on the PC board but are solidly affixed to the rear panel of the main chassis—a costlier but more reliable solution. Cosmetically the Aragon 18k also makes more of statement, although I find the snazzy, deeply grooved control knobs to be dirt-catchers as well as eye-catchers. The tape-monitor outputs are still paralleled, allowing a worst-case scenario of destructive, amplified oscillations (unlikely, yes, but not impossible).

The THD + N measurements show that the various design improvements have had only a minor effect. The distortion is very low indeed, but it was only 5 dB or so higher in the Acurus L10. The Aragon 18k reaches a minimum distortion of -98 dB at approximately 8 V out, with both a 20 Hz and a 1 kHz input. With a 20 kHz input some dynamic distortion is evident at output levels over 3.5 V, which is pretty academic. Minimum 20 kHz distortion is in the -90 to -93 dB range. The distortion curves are completely noise-dominated; the noise floor itself is about average for high-quality line-level preamps. A low-impedance load (600 ohms) has no effect on the maximum output level or minimum distortion. In channel separation the 18k surpasses the L10 by about 9 dB but only at the lowest frequencies, where the 18k bottoms out at 104 dB. At 20 kHz both models are around 73 dB.

It seems to me that Mondial Designs should be compared to one of those great Continental hotels where even the cheapest room is quite luxurious because they simply don't have the mentality to do it any other way. If you pay more you get more, but you don't need to pay more. The Aragon 18k is one of the better rooms.

Line-Level Preamplifier

Bryston BP20

(Reviewed by Peter Aczel)

Bryston Ltd., P.O. Box 2170, 677 Neal Drive, Peterborough, Ont., Canada K9J 7Y4.Voice: (705) 742-5325. Fax: (705) 742-0882. BP20 line-level preamplifier, $1395.00. Tested sample on loan from manufacturer.

David Rich, as an engineer, is one of the biggest fans of Chris Russell as an engineer, so David should have written this review of Chris's latest-and-greatest preamp. Because of various intramural errors—all of them my fault—David never had a chance to do so, and therefore, since "the buck stops here," I am closing up the breach.

The BP20 replaces the 11B/12B series in the Bryston line and represents a complete rethinking on the part of Chris Russell of just how a preamp should go together. The basic Bryston gain-stage topology, using operational amplifiers made up of discrete components and ±24 V power supplies, has not changed (at least as far as I can see), but the physical layout and construction of the unit are new and different. The main chassis is pancake-flat (barely 1½" high); the power-supply transformer is a separate external unit with an "umbilical cord." The PC boards have been redesigned, the signal paths simplified, and a ground plane incorporated. Each of the two channels has two XLR balanced inputs and one XLR balanced output, five unbalanced inputs and two paralleled unbalanced outputs, plus a tape loop. That provides a lot of interface flexibility. There is no record selector switch (a debatable signal-path simplification), only a toggle switch for Tape/Source selection. A balance control with 12 o'clock detent is next to the continuously variable volume control. The construction details are absolutely beautiful; parts quality is high.

Two favorable characteristics became apparent in the course of my measurements: (1) there was no "better" channel, the two being absolutely identical in distortion and noise, and (2) the balanced signal path was just as low in distortion and noise as the unbalanced. This uniformity was new in my experience. On the other hand, the balance control at the detent was off by 0.7 dB at unity gain with 2 V out (XLR in/out).

The THD + N versus level measurements yielded outstanding results. With either unbalanced in/out or XLR in/out, the 20 Hz and 1 kHz distortion dropped to a minimum of -97 dB shortly before the clipping level. The 20 kHz curve showed a small amount of dynamic distortion but only above 4 V and 8 V out (unbalanced and balanced, respectively), so it is of no significance. Maximum undistorted output was approximately 14 V unbalanced and 28 V balanced—I think that should be sufficient for any application, don't you? The curves were absolutely linear (i.e., noise-dominated); the noise floor appeared to be no better and no worse than I had seen in other topnotch preamplifiers, give or take a couple of dB. Channel separation, one of the stumbling blocks with a compact chassis and a balance control, was much improved over the 11B. Under worst-case conditions (balanced was worse than unbalanced, unity gain worse than full gain), the separation was 70 dB or better (up to 85 dB) at nearly all frequencies. Messing with the balance control made things worse at the highest frequencies; 54 dB at 20 kHz was the most screwed-up reading I was able to obtain that way. This is still highly acceptable performance.

The Bryston BP20 is only the second preamplifier we have tested—the first was the Krell KRC-2—that will accept both balanced and unbalanced inputs and deliver both balanced and unbalanced outputs. If that's what you need—and in certain highly elaborate audio systems you almost surely will—the Bryston outperforms the Krell in most respects at 38 cents on the dollar (unless you feel the Krell's CMOS remote control alone justifies the other 62 cents). As a purely unbalanced front end the Bryston has some rivals, though no indisputable superiors, and its look-and-feel make it a delight to use in any event.

Mono Power Amplifier

Marantz MA500

(Reviewed by Peter Aczel)

Marantz America, Inc., 440 Medinah Road, Roselle, IL 60172-2330. Voice: (708) 307-3100. Fax: (708) 307-2687. Model MA500 mono power amplifier, $299.00. Tested samples on loan from manufacturer.

This is so simple, it's simply beautiful. A single channel of "minimalist" low-distortion amplification without frills, on a deep and narrow chassis with a small frontal area, highly suitable for multichannel clustering in a small space, at a surprisingly low price. If this is what crass commercialism in audio engineering is all about, I want more of it.

The Marantz engineers went for what appears to be a genuinely smart performance vs. cost tradeoff here. The heart of the amplifier is the µPC 1342V chip, which incorporates all the voltage gain stages within a single IC. We have very little data on this highly "streamlined" chip. What we know is that it must see the full ±60 V power supply so it can drive the discrete output devices, which have no voltage gain. One generally wouldn't expect unusually high performance from a high-voltage monolithic chip, but the measurements of the MA500 are quite reassuring in this respect. The output stage of the amplifier uses dual paralleled discrete transistors. The coupling capacitors in the signal path are electrolytics (what else did you expect at this price?).

The dc rails are filtered by 10,000 µF capacitors. Overcurrent protection is almost identical to the system used in the considerably more costly Rotel RB-990BX. Overcurrent is sensed by a bipolar device, which is connected to a custom protection chip. The protection chip drives a relay, which is in series with the speaker terminal. In addition to overcurrent, the presence of a dc voltage at the output will also cause the relay to open, as will turn-on and turn-off. This level of protection is very rare at this price.

The MA500 is rated at 125 watts into 8 ohms and 180 watts into 4 ohms. Two MA500 units can be bridged to create a very high-powered amplifier channel rated at 450 watts into 8 ohms. THX certification is part of the package, guaranteeing among other things that 3.2 ohms is a viable load with continuous power. Any number of the amplifiers can be cascaded via their remote-control bus jacks and turned on/off simultaneously with the remote control of a Marantz front end (recent models only). All these little quality touches contradict the "cheap" image that might be suggested by the MA500's price. And, believe me, it doesn't *look* cheap, either.

My THD + N tests indicated that the specs are conservative. Clipping into 8 ohms takes place at 140 watts and into 4 ohms at just below 200 watts. The 20 Hz disortion dips to a minimum of -94 dB into 8 ohms and -91 dB into 4 ohms. The 1 kHz distortion is just 3 or 4 dB higher. The minima occur shortly before clipping, and the curves appear to be noise-dominated, with the noise floor itself unusually low. The 20 kHz curves into both 8 ohms and 4 ohms show some dynamic distortion, but even at clipping the THD stays below -70 dB and at reduced power levels is considerably lower (at 20 watts, for example, I measured -78 dB into 8 ohms and -73 dB into 4 ohms). These results are quite a bit superior to the performance obtainable with some very high-end power amplifiers, showing what is possible at minimal cost when real engineers are in charge. (Don't misunderstand or misquote me; this is still no Boulder or Bryston, but it's respectable.)

The PowerCube measurements showed very good dynamic power into 8, 4, and 2 ohms, regardless of the reactance of the load. At 0°, the intermittent 1 kHz tone burst maxed out at 40 V into 8 ohms, 36 V into 4 ohms, and 28 V at 2 ohms. At ±30° and ±60°, the readings were slightly higher, as they should be. The dynamic headroom at 8 ohms came to approximately 0.9 dB. I was unable to obtain reliable readings into 1 ohm because the protection circuit started to get nervous at that point. I'm not going to hold that against the MA500; such crazy low-impedance loads are strictly for expensive amplifiers with huge power supplies. At least the protection was there.

As you can probably gather from all of the above, this is my favorite "tourist-class" power amp so far.

Stereo Power Amplifier

PSE Studio IV

(Reviewed by David Rich)

Professional Systems Engineering, Inc., 9755 Hamilton Road, Eden Prairie, MN 55344-3424. Phone: (612) 943-1677. Studio IV stereo power amplifier, $995.00. Tested sample on loan from manufacturer.

The performance of a smaller power amplifier can often be superior to that of a larger amp because the drive requirements to the output device are reduced, thereby reducing predriver complexity. In addition, since the breakdown V_{CE} of the output devices is lower, it is possible to find a device with a higher unity-gain current frequency. Unfortunately, state-of-the-art designs are often reserved for high-power, higher-priced amplifiers.

The PSE Studio IV is a state-of-the-art design with a specified output of 100 watts per channel into 8 ohms. A two-transistor current source, biased by a zener-diode-based voltage reference, powers the input differential pair. The input differential pair is formed with degenerated JFETs. The differential pair is cascoded with bipolar devices and terminates to a resistive load. The second stage is a push-pull stage with resistive load. The total return loop gain is 40 dB, making this a moderate-feedback design. On the upper supply rail, the second gain stage is driven from an emitter follower. This important stage is often eliminated in El Cheapo designs. On the lower supply rail, the second gain stage is driven by a common-source amplifier degenerated to a gain of 1. Differential-to-single-ended conversion thus happens in the second gain stage. The second gain stage drives a MOSFET source follower, which in turn drives dual paralleled output stages. Biasing of the composite MOSFET/bipolar stage is difficult because of the differing temperature coefficients of the MOSFET and bipolar devices. (See the Parasound HCA-2200II review in Issue No. 21.) In the Studio IV, the biasing device is formed with an LED. The temperature coefficient of the LED apparently matches the temperature coefficient of the composite output stage. The output stage is current-limited by a bipolar-based circuit that is similar to the standard configuration but much more complex. This complexity is introduced in an attempt to prevent the circuit from activating into reactive loads when the current drain may be high but the voltage across the transistor is low enough so that the output device is still in the safe area of operation. A single channel uses 20 transistors and 12 diodes.

A single toroid and pair of large 27,000 µF filter caps form the unregulated supply rail for both channels. A dual mono design would have added to the cost of the unit, driving it beyond the $1000 price point. Nominal supply voltage is 50 V. The voltage gain stages are on a separate regulated 55 V supply. Higher supply voltage is required here as explained in the Parasound HCA-2200II review. Separate transformer windings and bridge rectifiers are used for the regulated rails. The regulators are formed with LM324 integrated regulators. The use of a regulated supply dramatically improves rejection of power-supply noise and is an important aspect of this design that many amplifiers costing much more than the Studio IV fail to include. With such a regulated supply the improvement in channel separation that a dual mono design could effect would be small. The measured channel separation was 68 dB at 20 kHz, 90 dB at 1 kHz, and 117 dB at 20 Hz. No coupling capacitors are used in the circuit. DC is nulled by a trim pot.

Construction is excellent. A large heat sink forms the front of the unit. Double-sided circuit boards are parallel to the heat sink to minimize wiring to the output devices. Some tweako signs in the construction include multiple-paralleled stranded wires used for power supply and output wiring, and a plastic top for the chassis.

Distortion would be expected to be higher than in the more expensive competition, owing to the lack of a cascoded second gain stage and the use of a simple, open-loop, MOSFET-based predriver stage. Measurements confirmed this. Into an 8-ohm resistive load, THD + N reached a minimum of -86 dB at 40 watts and then rose with a soft clipping profile to -78 dB at 83 watts. At the "official" 100-watt maximum output level the distortion was -58 dB (0.13%). These figures were almost the same at all frequencies; dynamic distortion was negligible. Into a 4-ohm resisitive load, minimum distortion was -83 dB, still at 40 watts, then rose with a soft clipping profile to -70 dB at 180 watts. Dynamic distortion was a little more evident into 4 ohms, especially in one channel, where the 20 kHz curve deviated from the 1 kHz curve by a worst-case margin of 7 dB. (The MOSFET/bipolar output stage may have a bias-tracking problem that causes this variation.) The inherent noise floor of the amplifier was significantly lower than that of some prestigious high-end units.

The PowerCube system measured a dynamic output voltage of 33 V (136 watts) into 8 ohms. That represents a dynamic headroom of approximately 1 dB, as referenced to the steady-state output of 110 watts at 1% distortion. The PowerCube looked outstandingly good—almost ideal—with both resisitive and highly reactive loads at 8 ohms and 4 ohms, but at 2 ohms and 1 ohm the output dropped drastically into reactive loads. This indicates that the protection circuit is coming on despite its complexity. Into pure resistance the output was still good at 2 ohms and 1 ohm, down 15.5% and 41.4% respectively. Peak current output was 37 amps.

Given the complexity of the circuit design and the relatively modest power output of this amplifier, one would think it had the potential to perform at the Boulder or Bryston level. The PSE Studio IV doesn't quite live up to that potential. Do not get me wrong; it performs as well as other amplifiers we have recommended. Unfortunately, one such amplifier is the Marantz MA500 (see the Editor's review above), which costs significantly less and offers slightly more power into an 8-ohm load. For the additional $397, the American-designed and manufactured PSE gives you better build quality and better power supply rejection, owing to its regulated supplies. For many this will be enough to justify its cost, and I would not discourage them from purchasing this unit. But we might think of the PSE Studio IV as work in progress because, with small modifications to the protection circuitry, it would draw an excellent PowerCube, and changing the output-stage predriver from MOSFETs to bipolars could probably bring the distortion down by about 10 dB. The Studio IV would then be able to challenge directly its more expensive competitors.

Line-Level Preamplifier

Rotel RHA-10

(continued from Issue No. 21)

(Reviewed by Peter Aczel)

Rotel of America, P.O. Box 8, North Reading, MA 01864-0008. Voice: (800) 370-3741. Fax: (508) 664-4109. RHA-10 Stereo Active Controller, $1799.90. Tested sample on loan from manufacturer.

This is supposed to be the high-end predecessor of the modestly priced RC-980BX, and there are some resemblances but they are generally unflattering to the RHA-10. Unlike the cheaper unit, the RHA-10 uses discrete transistors in the gain block. Complementary bipolar differential pairs are biased by single-transistor current sources and resistively loaded. The second gain stage is a push-pull degenerated bipolar circuit, which drives a push-pull emitter follower. The second stage is not resistively loaded. Referring to the generic active amplifier stage explained on page 18 of Issue No. 18, the coupling capacitors C_1 and C_3 are electrolytics, just as in the cheaper Rotel, only bigger (25 µF instead of 10 µF). The more important C_2 is not used, again as in the cheaper unit.

As for the power supply, each channel has its own bridge rectifier and open-loop regulator. The pass transistor regulator is of a two-transistor composite design and is set to the regulated voltage by a zener diode biased by a current source. Using the same zener reference for both channels here is a peculiarly chintzy feature in an expensive preamp, more understandable in the similarly configured RC-980BX. The power transformer, on the other hand, is large for a preamp.

As in the cheaper unit, the output stages are muted by relays on power-up. Unlike the cheaper unit, the RHA-10 has its tape monitor outputs buffered by AD712 op-amps. The record selector is correctly designed to prevent self-looping of a tape recorder, which could lead to destructive amplified oscillations. The input and record selector switches, as well as the volume control, are sealed units. The volume control is a motorized pot, of good quality but not on the Penny & Giles level, controllable with the remote that also controls the Rotel RHT-10 FM stereo tuner. The PC boards are double-sided, but the RCA jacks are mounted directly on the board, another chintzy touch in this price range. The sheet metal is cosmetically more attractive than the cheaper unit's but not really of superior quality. The screws are soft and too easily strippable.

Although the basic engineering of the RHA-10 is quite ordinary and in some details downright parsimonious, careful execution has resulted in extraordinarily high performance. Would you believe -100 dB THD + N at 7 V out? At any frequeny, including 20 kHz, with hardly a trace of dynamic distortion? That's the best we've measured so far. The noise floor, however, is not lower than in other high-quality preamps. Maximum output before clipping is just below 14 V. A low-impedance (600 ohm) load reduces maximum output to 12 V and causes some dynamic distortion above 2.5 V out. Channel separation is excellent; at unity gain it ranges from 105 dB at the lower frequencies to 94 dB at 20 kHz. That't partly due to the dual volume control in lieu of a separate balance control.

Overall, the Rotel RHA-10 is a damn good unit, but I just don't see $1800's worth of preamp under the cover. At $900 or even $1000 I'd be wild about it.

Stereo Power Amplifier

Rotel RHB-10

(continued from Issue No. 21)

(Reviewed by Peter Aczel)

Rotel of America, P.O. Box 8, North Reading, MA 01864-0008. Voice: (800) 370-3741. Fax: (508) 664-4109. RHB-10 stereo power amplifier, $2699.90. Tested sample on loan from manufacturer.

After further examination, reflection, tests, and headscratching, I am not at all enthusiastic about this unit and don't feel like giving it much space here. Not that it isn't an excellent power amplifier. It is. But it costs 2½ times as much as the Rotel RB-990BX and is virtually the same amplifier! The circuit is almost exactly the same, except for an emitter follower after the differential pair and before the second gain stage. The output-stage transistors are of a different type. The heat sinks are larger but not much. There are two power transformers, not a single big one, and more full-wave rectifiers and filter caps—but still no regulated supplies! The PC boards are still single-sided—hey, at $2700? Protection is better, with relays in series with the outputs. The sheet metal is prettier but not greatly superior. I just don't see the $1600's worth of extras. Yes, the power output is a little higher (210 watts per channel into 8 ohms, 420 watts into 4 ohms) and the THD + N lower (minimum around -95 dB, with plenty of dynamic distortion at 20 kHz). That's not enough to light my fire at this price. Sorry.

Passive Control Unit

Rotel RHC-10

(Reviewed by Peter Aczel)

Rotel of America, P.O. Box 8, North Reading, MA 01864-0008. Voice: (800) 370-3741. Fax: (508) 664-4109. RHC-10 Passive Controller, $999.90. Tested sample on loan from manufacturer.

As Gertrude Stein might have said, had she been an audiophile, "a passive front end is a passive front end is a passive front end." What's there to review about it? Well, maybe parts quality and construction. Crosstalk, yes. But harmonic distortion? Or, especially, the sound? Only the untutored cultists at the tweako magazines indulge in that kind of rubbish. A passive front end is a bunch of jacks, wires, switches, potentiometers, and suchlike. Might as well talk about the audio performance of the light switch in your wall. (Of course, the tweaks *do.)*

All right, the parts quality in the RHC-10 is high. The construction is beautiful. The crosstalk is remarkably low. Under absolute worst-case conditions, I measured better than 60 dB channel separation even at 20 kHz, improving by several orders of magnitude depending on frequency, attenuator setting, and source impedance. You can say that the unit is transparent.

The Input switch has 5 positions, corresponding to 5 pairs of unbalanced input jacks; the Rec Out switch has 4 positions plus Off. The outputs are Main Out and Record Out, unbalanced. The continuously variable high-quality attenuator has concentric L and R knobs, adjustable separately for balance and friction-locked for volume.

I don't see much point in passive front ends because well-engineered and correctly buffered active units are equally transparent and more versatile, but if the idea appeals to you and you think $1000 is chicken feed, this is a very fine passive control, recommended without reservations. Remember, though, that it has no gain and a 10k output impedance. Use only short interconnects.

Stereo Power Amplifier

Sunfire (more a preview than a review)

(By Peter Aczel and David Rich)

Sunfire Corporation, P.O. Box 1589, Snohomish, WA 98290. Sunfire "Load Invariant High Fidelity Stereo Power Amplifier," $2175.00. Tested sample on loan from manufacturer.

Here is another instance of being able to report an exciting new development thanks to our habitual lateness. We were able to have Bob Carver's superduper hot-off-the-production-line Sunfire amplifier for about 24 hours and run a few tests on it before (finally!) going to press.

First, let's get all the identities straight. Sunfire Corporation is the new, small, high-end company Bob Carver formed after he had left Carver Corporation. (That breakup was widely reported in the business press, so I won't go into its Byzantine intrigues and details. Bob is still the largest, though not majority, stockholder of Carver Corporation, but without a job there.) The Carver company also has a new amplifier, called Lightstar and in many ways similar, but not identical, to the Sunfire. It was started by Bob before he left and finished by others. We have been promised a sample of the Lightstar, but this preview is of the Sunfire, i.e., the all-Bob-Carver execution of the same basic design concept. (Lightstar, Sunfire... Guess who made up those names?)

The amplifier is very handsome in a sleek, black, understated way; much more High End in appearance than any of the older Carver models; beautifully constructed with high-quality parts—and a joule meter as the pilot light! It has two outputs per channel, one with very low output impedance (voltage source) and one with a huge 1-ohm series resistor (current source). The latter simulates a typical tube-amplifier output, yielding a softening of the highs and a slight plumping of the lower frequencies when feeding a typical speaker load (see Issue No. 16, page 55). I don't believe that such "processing" belongs in a power amp; in fact I don't believe in tube amplifiers with highish output impedances; but of course you don't have to use that output.

I had time to make only a few measurements and I want to discuss the results only in a very general way, until we receive our "official" review sample and can test it unhurriedly. The Sunfire is a 700-watt-per-channel stereo power amplifier. That's a continuous-power measurement into 4 ohms, just below clipping, at any frequency. Is that enough muscle for you? With intermittent 1 kHz tone bursts as the input, it can deliver over 2000 watts per channel into 1 ohm! What's more, it doesn't care whether the load is resistive, capacitive, or inductive. It just keeps pumping almost constant voltage into any phase angle. The exception is 1 ohm at -60° and +60°, where the output voltage drops considerably—but no loudspeaker presents that kind of load. Into 1 ohm at -30° and +30° the output voltage is still remarkably high. The amplifier is labeled "load-invariant" on the fascia, and that's a fair statement, at least at the present stage of our investigations. Our usual THD + N measurements include harmonics and noise up to 80 kHz; in the case of the Sunfire this test is skewed by the high-frequency switching noise. Reducing the measurement bandwidth to 22 kHz yields dramatically better readings, and with an A-weighting filter there is another huge improvement. A detailed discussion of the Sunfire's measured performance will have to wait until we receive a sample we can keep for a while. Meanwhile, my initial impression is that this is an amplifier of great originality and unique capabilities.

—Peter Aczel

To borrow from NPR's "Classical Countdown," this amplifier is genuinely interesting. The problem that Bob Carver is addressing is the power dissipation in a power amplifier that occurs as the amplifier drives the loudspeaker. As we explained in Issue No. 20, this power dissipation is the result of the amplifier's finite conversion efficiency. One method to improve conversion efficiency is to use class D (switching) power amplifier design. These circuits are very efficient because the unregulated power rails are often switched off from the load. The problem with switching amplifiers is that the switching clock (100 kHz and above) can leak into the output, violating FCC regulations. In addition, the design of a low-distortion PWM amplifier can be very difficult, as has been demonstrated in numerous AES preprints.

What Sunfire does is to use a switching amplifier to generate the power supplies for a conventional amplifier. The switching power supplies are set to be ±7 V away from the output signal level. The amplifier itself is a conventional design which can be made to achieve as good performance as desired using standard techniques. Switching noise is attenuated by the power-supply rejection of the amplifier. Distortion in the power-supply rails has almost no effect on the amplifier, even if it is 10%. So we have almost the efficiency of a class D amplifier but without the performance problems associated with that technology.

The extra pair of tracking/switching power supplies adds a lot of electronics to the amplifier but does not add to the cost. The reason is that the large chassis and heat sinks normally required to dissipate the heat of a high-power amplifier designed to drive low-impedance loads is eliminated. While it may come as a surprise to many of you, it turns out that sheet metal costs a lot more than electronics, so in the end Sunfire can produce the amp at the same cost as a comparable conventional design, or at even lower cost. What is gained is that with its very high efficiency the amplifier can drive very low-impedance loads continuously. Even the largest of the conventional designs cannot drive loads of 2 ohms and below continu-

(continued on page 49)

Catching Up on the Digital Scene

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

Some of these reviews were left out of the last issue; some are new; all are still of current interest.

We were hoping to bring you in this issue a definitive article by Steve Norsworthy on the pros and cons of delta-sigma D/A converters, explaining idle channel tones, the proper use of dither, etc., but Steve got totally tied up in projects of higher priority (to him, that is). The article *is* coming, though, and when it comes it should have the same kind of impact as Bob Adams's jitter article in the last issue. (The general reaction to the latter was a sigh of relief that an authoritative, professional, yet relatively simple explanation of a tweako-cultist-ridden subject was at last available.)

Compact Disc Player

Denon DCD-2700

(Reviewed by David Rich)

Denon America, Inc., P.O. Box 5370, Parsippany, NJ 07054-5370. Voice: (201) 575-7810. Fax: (201) 575-2532. DCD-2700 compact disc player with remote control, $1200.00. Tested sample on loan from manufacturer.

Editors Note: *Since this review was written, the DCD-2700 has been succeeded by the DCD-3000, at the same price. No service manual for the newer player, or even a circuit schematic, was available from Denon as of press time, only the operating instructions, from which the differences appear to be relatively unimportant, without probable bearing on the major points made by Dr. Rich. The transport has been moved to the center of the chassis and is of a slightly different design; digital inputs have been added, allowing the use of the newer player as an outboard D/A processor (but would you want to do that?); balanced analog outputs have been added; the pitch control feature has been eliminated; the weight of the player has been reduced by 13%. Since the dollar buys fewer yen today than a year ago, I don't expect the newer player to be more expensively made;* au contraire. *The centerpiece of the design, which is the Alpha System processor, has not changed, and that alone makes the review that follows still fully relevant.*

* * *

This CD player is for all practical purposes Denon's top of the line. The company does produce a five-figure, two-piece transport/processor setup, but it is sold in very limited distribution. The DCD-2700 is loaded with all the standard features found at this price point, including a motorized volume control. The most unusual feature from the user's point of view is the pitch control function. *[Eliminated in the DCD-3000, as noted above.—Ed.]* A pair of buttons on the unit are pressed to raise or lower the playing speed. The unit also has a peak search feature for setting the tape-deck record level. Peak search scans the CD for the maximum signal level on the disc and plays a few seconds on either side of this point. Going to a specific index point is more difficult on this unit than most. (How strange, since Denon CDs use index points more than anybody else. Denon indexes all important subsections of a composition). You have to press the index button and then punch in the track and index numbers from the keypad.

The most surprising thing about the operation of this CD player is how long it takes to access tracks. It takes twice as long as one would expect for a unit in this price class with a linear drive transport. Indeed, access time is comparable to that of the Marantz CD-63 and Harman Kardon HD7725 CD players, which use the El Cheapo gear-driven laser mechanism. To find out what's going on we have to remove the cover. The first thing you see is a structural reinforcement bar that runs from the front to the back of the unit. It is held in place by friction only. Remove it and the reason for the player's slow tracking becomes apparent. The laser transport mechanism is a low-end gear-driven unit, just like those in the Harman Kardon and Marantz players!! This type of cost cutting is not what the purchaser of a $1200 unit should expect. The only interesting thing about the mechanism is that the laser pickup (Sony KSS-240A) has the RF amp built in. This makes the laser pickup less susceptible to electrostatic damage. The servo circuit uses an analog technology instead of the newer (and probably cheaper to manufacture but not necessarily better) DSP-based servo circuit used in the Sony and Marantz CD players. *[The DCD-3000 has a digital tracking servo system.—Ed]*

The transport is not the only sign of cost cutting. The PC board is a single-sided affair with lots of jumpers. A single smallish power transformer is used. It is similar in size to the one in the Marantz CD-63. Two full-wave rectifiers are used in the power supply, one for the digital circuits and one for the analog. Unregulated filter capacitors are only 2200 µF for the analog side (by comparison, the Harman Kardon HD7725 uses 4700 µF caps on the analog side). The rated working voltage for all the electrolytics in the analog supply, including these capacitors, is a surprisingly high 50 V (all is not cost cutting on this unit). A single regulator (7805) and two overcurrent protection devices are all the supply regulation that's on the digital side. On the analog side a pair of regulators, 7812 and 7912 (±12 V supplies for $1200—how come Marantz does it for $399?), are all that's used. The regulators supply both channels. All regulators are mounted on the chassis for dissipation of heat. The DACs are subregulated with 78L05 and 79L05 regulators (the L is for low current—translated: low cost). Separate regulators are used for the left and right channels on the DAC. Spectral plots of the player's line output showed significant hum components of 60 Hz and 180 Hz, peaking at approximately -115 dB relative to full scale. CD players in this price range usually do better, given their more robust power supplies. On the other hand, channel separation was excellent, 120 dB at 1 kHz and 100 dB at 20 kHz, so clearly signals are not crosstalking through the power supply.

The digital filter chip used in this unit is the NPC SM5845AF. Here Denon spares no expense. Many high-end manufacturers of five-figure units claim the much less expensive SM5813 is just perfect, but NPC apparently disagrees since they produced this more complex chip. Denon states that the SM5845AF is essentially the SM5842AP with an added feature discussed below. The SM5842AP has a data path which is very wide at 24 bits; coefficient word length is 26 bits, and the accumulator is 32 bits wide. The number of taps in the FIR filter was not specified. Stopband rejection is excellent at -117 dB, exceeding even Sony's current best effort, the CXD2567. The passband ripple of the SM5842 is ±0.00002 dB (here Sony wins by 0.00001 dB). The chip has a built-in dither generator to decorrelate the signal from the truncation error that occurs when the accumulator output is truncated to 20 bits. It also has digital de-emphasis. Strangely, we measured a peak of 0.08 dB at 4kHz when measuring the DCD-2700's frequency response in the de-emphasis mode. Even stranger are some 0.02 dB ripples in the passband frequency response of the DCD-2700 from 2 kHz to 13 kHz. These measurements make me wonder if something has gone wrong in the evolution of the SM5842AP into the SM5845AF.

The SM5845AF adds digital circuitry which attempts to estimate the value of a quantized data point at a resolution below the quantization level. This can only be accomplished by making some estimate of the type of signal that was digitized (because Dr. Shannon has a little theorem forming the basis of information theory that says you cannot take a quantized PCM signal at the Nyquist rate and extract any more information from it). When you play undithered low-level signals through the Denon, the outputs are much more continuous in appearance, and distortion of the signal is lower, than the theoretical minimum for an undithered signal. But a quantized sine wave *should* have distortion. When dither is not used, the quantization error is correlated with the signal and this causes the distortion. *It is for this reason that all recordings are done with dither before the A/D.* Dither added at the recording site decorrelates the quantization error, and the distortion is eliminated. Noise, not distortion, occurs instead. Dither also allows us to record signals at levels below an LSB. The Denon digital filter produces a low-distortion undithered sine wave, but how can it improve things with dithered music signals? To do this it would have to reduce the *noise* level of the signal. Since the noise is decorrelated with the signal and the filter chip has no way of estimating the statistics of the dither or the music, I do not see how this can work. It looks to me as if the whole thing were designed to make undithered sine waves look good at hi-fi-show demos, but Denon's information on this technique—they call it Alpha System Processing—is very sketchy, and a small possibility exists that the Denon system could be of some value. Audio Alchemy is claiming to have a device which performs a similar function, so one must be very careful not to dismiss all this without more information.

The DACs are of the Burr-Brown PCM1702J type. The PCM 1702 is a device which is a redesigned PCM63 in a BiCMOS process. It is a smaller die that fits into a 16-pin DIP package or an even smaller surface-mount package. The pins eliminated include the pins for pot adjustments, which were virtually impossible to adjust correctly. The spec sheets for both devices are similar, but the THD for the K grade of the smaller type is 2 dB worse at -20dB and dynamic range is no longer given a worst-case spec. Minimum idle-channel SNR is 6 dB worse at a still excellent 110 dB, and the PSRR spec is not given. Worst-case gain linearity is not specified and the typical number has changed from ±0.3 dB to ±0.5 dB at -90 dB. On the DCD-2700 we measured a gain linearity error of +0.25 dB at an input signal level of -80 dB. This then rose to +0.75 dB at -90 dB and +1.25 dB at

-100 dB. The CBS test disc monotonicity test looked excellent.

Two DACs are used per channel in a balanced configuration. The balanced signals come from the SM5845AF digital filter (a feature not found on the SM5842AP). This approach is less effective than when used with 1-bit coders because the 1-bit coders' outputs are highly repeatable from chip to chip since they are almost totally digital. The analog *R-2R* ladder-based DAC chips should not exhibit significantly correlated nonlinearities (see Issue No. 15). I would rather have a single pair of K-grade DACs.

The ICs in the analog path are of the NEC µPC4570 type, one chip per DAC (4 total). The µPC4570 is also used for the balanced-to-single-ended converter and the buffer for the GIC filter that drives the output terminals. Although the µPC4570 is a dual device, only a single op-amp is used; the other op-amp is not used (don't look at me, I don't know why). The NEC µPC4570 is said to be an NE5532 equivalent in the NEC data sheet and the specifications are similar, but the parts are not the same. The principal specification difference relates to the output stage: the NE5532 is designed to drive 600 ohms; the µPC4570 can drive only 2000 ohms. The µPC4570 uses a simple 2-gain-stage topology; the NE5532 uses a more complex 3-gain-stage topology. The simpler circuit and smaller output stage make possible a smaller die and a lower-cost part. For $1200 I would expect a state-of-the-art op-amp such as the AD797, but instead you get a cost-reduced NE5532.

But the worst is yet to come.

The generalized impedance converter uses an NJN4558 from the New Japan Radio Company (JRC). From the data sheet: "...combining the features of the NJM741 with the close parameter matching and tracking of a dual device." Yes, folks, this is a dual version of the 25-year-old 741!! Slew rate, for those of you who forgot (normally this thing is not used in audio even in the cheapest of the cheap because it is not fast enough), is 1 V/µs and the gain-bandwidth product is 2 MHz. What happens when you put a slow op-amp in a GIC filter with a 30 kHz passband edge frequency? You get Q enhancement that results in the peaking of the filter's response. [Sedra and Brackett. *Filter Theory and Design: Active and Passive,* Chapter 9.8.] That's just what we measured. The frequency response of the unit stays in a ±0.05 dB window (those strange ripples discussed above) until 13 kHz, then it rises up to +0.35 dB at 20 kHz. I ran a computer simulation of the circuit, using a macro model of the NJN4558, just to make sure that the op-amp was the cause of the peaking. The simulation results corresponded closely to the measured results, but substituting an ideal op-amp in the simulation reduced the peaking by only 0.2 dB. It turns out that the remaining peaking is caused by the use of 5% carbon resistors throughout the analog signal path, instead of 1% metal film resistors that we expect to find in a four-figure-priced unit. Since 5% resistors are available in a much more limited number of resistor values, the exact values required for the filter are unavailable, and the values actually used cause the remaining peaking. All I can say is, the person who designed this stage must have come straight from the design of a $100 portable CD player, and he/she/it (I hate this PC stuff) did not change his/her/its design methodology for this $1200 unit.

Full-scale THD + N was substandard at high frequencies. It measured -95 dB at 500 Hz and then it started to rise. Measuring -90 dB at 3 kHz, it rose to a maximum of -63dB (!!) at 13 kHz. Dropping the input level to -24 dB resulted in distortion readings of 1 to 2 dB above theoretical limits. This indicates the distortion is coming form the analog stages and is not unexpected, given the analysis of the analog stage above.

The dc blocking capacitor at the output of the analog stage consists of a series-parallel quad of electrolytics. Each series combination has the plus terminal of the capacitors tied together. This is an attempt to mimic a nonpolar capacitor. One set of capacitors is 10 µF, the other is 470 µF. One sign of quality design is the muting circuit, which uses a relay that shorts the output to ground.

Summing it up:

This unit is overpriced, given the parts inside. Some interesting aspects of the design are the balanced DAC and the digital filter, but the analog section is a poor performer (if I were Corey Greenberg, I would have a more colorful way of putting this). Denon has produced products that performed well and represented good value in the past. I hope the DCD-2700 is an aberration.

Outboard D/A Converter with Transport

DPA (Deltec Precision Audio) PDM 2 and Tl

(Reviewed by Peter Aczel)

Audiophile Imports, 2012-B Main Street, Cross Plains, WI 53528. Voice: (608) 798-3338. Fax: (608) 798-3359. DPA (Deltec Precision Audio) PDM 2 Bitstream D/A converter, $4395.00; T1 transport with remote control, $1595.00. Tested samples on loan from distributor.

This U.K. company, which sometimes signs itself as "dpa digital ltd" and has its headquarters in Wales, makes ultrahigh-end digital processors. The costly equipment reviewed here is not even their top-of-the-line at this point, although it was at the time we tested it and is still being sold at the above prices, which are—believe it or not—reduced from the original. The designer of the circuits and managing director of the company is Robert Watts.

The juxtaposition of the PDM 2 D/A converter to the Tl transport is a somewhat schizoid one because the two-chassis PDM 2 is exceedingly high-tech, whereas the Tl is definitely not. To dispose of the Tl first, it's a cheap plastic Philips player with what David Rich called a "Hollywood false front"—namely a machined aluminum front panel and a metal outer skin with rounded edges and a high-gloss black finish. (The two PDM 2 units are similarly packaged, only much smaller.) The transport doesn't have the all the sophisticated disc-control facilities one expects these days in a high-priced player, but the interface with the PDM 2 is as sophisticated as it gets. DPA actually had to chop off some of the original Philips plastic to make room for the interface board and its fair-sized transformer.

The rest of the system is incredibly advanced in some respects and old hat in others. The analog and digital sections of the PDM 2 are in two entirely separate boxes. The Philips DAC 7 (TDA1547) Bitstream chip is used as the D/A converter, in conjunction with the Philips SAA7350 Bitstream PDM device, but the implementation is much more elaborate than in any other processor we have tested. Most of the components are surface-mounted, a technology almost never seen in low-volume high-end audio equipment. It eliminates the leads, gets rid of parasitic capacitance, allows a large number of components in a small circuit-board area, and saves a tremendous amount of space. Only wave-soldered, double-sided PC boards are used, and they're really packed. Each channel has a custom hybrid circuit in the signal path, another ultrahigh-tech feature hardly ever seen in consumer electronics. (A hybrid circuit permits the mixing of discrete and IC technologies of every type, as well as great flexibility in resistor and capacitor values, tolerances, etc.) On the other hand, the S/PDIF decoder is the mundane, old-tech Yamaha YM3623B, with no signs of any auxiliary circuitry, and the digital filter is the similarly banal Yamaha YM3414. On the plus side again, the switches are high-quality sealed units, the muting and de-emphasis switching is by relays (not solid-state switches), and both the analog and digital power supplies appear to be quite elaborate and well-regulated.

A multiplicity of fiber-optic cables is used to connect the digital box to the analog box. These Toslink connections carry very high-speed signals transmitted at a high oversampled rate, too fast for a single plastic optical cable to transmit. This is basically an isolation technique, designed to prevent intermodulation of the digital and analog signals. An additional fiber-optic cable sends the DAC clock back to the CD transport, thereby assuring clock synchronicity. DPA calls this the "Deltran Sync-Lock," although Sony used the same jitter-elimination technique a good many years ago in their high-end CDP-Rl and DAS-Rl—but that doesn't make it a less good solution.

On the laboratory bench the DPA system yielded mixed results. Full-scale THD + N was unimpressive; at the lower frequencies it hovered in the vicinity of -85 dB, rising to a maximum of -82 dB at 7 kHz. These were the figures with an S/PDIF input; with a CD input from the Tl they were worse by 2 to 4 dB. Reducing the input signal level in order to get rid of gain-related analog distortion still did not improve the results sufficiently; the best obtainable figure as normalized to full scale was -95 dB, still 3 dB short of the theoretical 16-bit minimum, but at least the worsening at the higher frequencies was gone. The DPA literature shows an FFT plot of a dithered 1 kHz tone at -60 dB to prove that the noise floor is an astonishing -130 dB; I was able to duplicate this, but the plot is deceptive—the distortion/noise is shown bin by bin, not as an rms total. Other processors I have measured can produce an equally good plot under the same conditions. The noise floor as measured conventionally with digital zero input is actually not so great; there are power-supply bumps at 60 Hz and 180 Hz hovering around the -100 dB level, and the noise reaches -97 dB at 60 to 90 kHz. Repeating the FFT test at full scale or even -10 dB with a dithered 7 kHz tone showed some very peculiar noise-floor modulation effects; Steven Norsworthy and David Rich are still debating the cause. It can't possibly be a good thing. On the other hand, gain linearity was of the utmost perfection, with zero error all the way down to -100 dB (that's delta-sigma for you). The full-scale frequency response of the PDM 2 drooped to almost -1 dB at 20 kHz; I wonder if that's really necessary. At 10 kHz the response was -0.25 dB. A very slight audible softening of the highs is therefore a possibility. I didn't agonize over it when I listened; the sound was excellent, needless to say.

Speaking of Norsworthy and Rich, their recent AES paper on idle channel tones in delta-sigma converters involved proprietary test techniques that were tried on the PDM 2 among many other processors, and it appears that the complete separation of the analog and digital electronics pays off handsomely in this particular respect. The PDM 2 produced significantly smaller amounts of idle channel tones than other processors using the same Philips chip set. A detailed explanation of how and why this happens will have to wait until the groundwork is laid by our scheduled introductory article on the subject.

Bottom line: this DPA equipment incorporates some very advanced technology, but on a black-box basis a number of other designs perform better. The latest DPA models, which are just out, may be a different story.

Compact Disc Player

Enlightened Audio Designs CD-1000

(Reviewed by Peter Aczel)

Enlightened Audio Designs Corp., 300 West Lowe, Fairfield, IA 52556. Voice: (515) 472-4312. Fax: (515) 472-3566. CD-1000

"Audiophile CD Player" with remote control, $1599.00. Tested sample on loan from manufacturer.

I can make short shrift of this one because I have in effect reviewed it before, though not in this form. The transport uses the offbeat but well-designed Pioneer "Stable Platter" upside-down drive mechanism, which I reviewed as part of the Pioneer Elite PD-75 CD player in Issue No. 18. The electronics are essentially the same as in the Enlightened Audio Designs DSP-1000 Series II outboard D/A processor, which I reviewed in Issue No. 20 and recommended as an excellent choice. (In fact, David Rich bought one.) I could end my review right there, leaving you with the impression that this is a highly desirable CD player, but there's a little more to it than that.

For one thing, the PD-75 (which was originally introduced in 1991 and is no longer in the Pioneer line) had index buttons on its remote control; the EAD CD-1000 does not. That alone would be sufficient for me, personally, not to buy it even at half the price, but those who have few or no classical CDs probably won't mind (rock-pop CDs never—or hardly ever?—have their numbered tracks further divided into index numbers). I am firmly opposed to the use of the forward and backward scan buttons for index hunting; it's a drag (and let's not even talk about the CBS CD-I standard test disc, which is all indexes). Other than that, the transport of the CD-1000 is just fine and dandy (see my comments in the aforesaid PD-75 review).

I expected the standard measurements to show pretty much the same results as with the EAD DSP-1000 and I was at least partly right. In the right channel, THD + N at full scale (0 dB) was 5 dB worse than the theoretical ideal (-98.08 dB) at nearly all frequencies. That was clearly due to gain-related analog distortion because, with the digital signal level reduced from 0 dB to -24 dB, the departure from perfection was only 1 dB, matching the performance of the DSP-1000. The left channel, however, was another story. At full scale, the excess distortion was 11 dB at the lowest frequencies and never better than 6 dB at any frequency. Even at -24 dB, there was still approximately 3 dB excess distortion at all frequencies—much worse than the other channel. I suspected low-level hum components (60 Hz and multiples), but spectrum analysis revealed no such bumps, only a generally elevated noise floor, especially between 90 and 500 Hz, about 10 dB higher than in the right channel. David Rich suspects a noisy op-amp. I'll give EAD the benefit of the doubt and assume that other samples of the CD-1000 do not have this fault, but QC should have spotted it and weeded it out before the unit was released.

In all other respects the CD-1000 proved to be flawless on the lab bench. At full scale, the frequency response rolled off 0.05 dB at 10 kHz and 0.15 dB at 20 kHz. De-emphasis error was 0.15 dB at 10 kHz and 0.35 dB at 16 kHz, but that included the frequency response decline. Channel separation was 86 dB at 16 kHz and improved 6 dB per octave downward. Gain linearity error down to the -100 dB level was virtually zero (remember, the DAC here is the 20-bit Burr-Brown PCM63P-K, as in the DSP-1000), but the low-level noise in the left channel was clearly worse in this test, also. Harmonic distortion of a dithered 997 Hz tone at -90.31 dB was just about undetectable above the noise floor.

If it had not been for the low-level noise problem in the left channel and the lack of index buttons on the remote control, I would have considered the EAD CD-1000 to be a close runner-up to the Sony CDP-X707ES, at a saving of $401, as my favorite "Audiophile CD Player" (to use the phrase inscribed on the EAD's front panel). As it is, I cannot give it that ranking. But, if EAD came out with a CD-1000 Series II…

Compact Disc Player

Marantz CD-63 and CD-63SE

(Reviewed by David Rich)

Marantz USA, 1150 Feehanville Drive, Mount Prospect, IL 60056. Voice: (708) 299-4000. Fax: (708) 299-4005. Model CD-63 compact disc player with remote control, $399.00. Model CD-63SE compact disc player with remote control, $499.00. Tested samples on loan from manufacturer.

Editor's Note: *Although this review was written in the spring of 1994, the CD-63 is still in the Marantz line and will remain therefor some time. Meanwhile, Marantz has also added the CD-63 Special Edition to the line. The SE version is listed at $100 more and is basically a marketing idea, designed to attract those who like the idea of "tweaked" versions of standard components. At least the tweaking here is done by the designers of the original unit, not by some third-party cultist operating from his basement. The "improvements" include a much heavier bottom plate for the chassis (the SE weighs 37% more than the plain CD-63), gold-plated jacks, a number of fancier capacitors, a higher-grade power transformer, additional shielding, etc. All very nice, but I could measure no improvements of any kind in the actual output of the SE. The two models are indistinguishable in performance—and that includes the small power-supply bumps rising from the noise floor. Thus every word of Dr. Rich's review is applicable to both models.*

* * *

The marketing guys at Denon and Harman Kardon, having read my reviews of their CD players, are no doubt saying something like this: "That Dr. Rich is some ivory-tower fruitcake. If he knew anything about the cost of producing a CD player, he would understand that you can't produce a CD player the way he wants to at our price point." To these marketing people I answer: Marantz CD-63. Now do not get me wrong, the Marantz CD-63 is not a Sony CDP-X707ES for one fifth the price—far from it. But its parts/price ratio is much better than we have seen in the Denon and Harman Kardon units. You even get a coax digital output in addition to the Toslink. This is very rare at this price point. The unit certainly does not look like it's from the bargain basement, although the disc-search switches feel pretty clunky. Only one important feature is missing but it's a big one. The unit does not display indexes and it has no provision to go to an index point on the disc. *[The SE version does not remedy this serious shortcoming.—Ed.]*

The unit has an all-metal cabinet. Plastic is very common at this price point. The power transformer is about the same size as that of the Denon DCD-2700. The CD-63 has the same number of regulators (4) and the same analog supply rail (±12 V). Such large supply rails are very rare at this price point. To be fair to the Denon, it has larger filter capacitors and additional DAC subregulators. But the Denon has a single-sided PC board, whereas the Marantz uses a double-sided board so a ground plane can be added. (There are no plated-through holes on the Marantz board, so it is still full of jumpers.)

A digital signal-processor chip is used for the servo functions (TDA1301). This is also the way the Sony CDP-X707ES does it, but they have developed their own chip (CXD2501Q). The actual transport is almost literally a toy in comparison with the Sony's. It's almost all plastic. According to the Philips literature, it's "designed both for stationary use and for application in CD-Radio Cassette Recorders....The low mass actuator is particularly effective in low power and portable applications." Low-profile design is again aimed at the portable market. A cheap-looking rubber suspension looks as if it were not going to work as well as a spring-loaded floating metal subframe, such as used by Denon, Harman Kardon, and Sony, but an additional plastic subassembly surrounding the transport and apparently filled with a viscous substance provides additional isolation. The build quality of the transport and chassis is what justifies the Sony CDP-X707ES to be five times the price. The Sony would have to last five times as long to recover the cost difference between the two players. I do not think it is going to last that long, but there is no question that you get what you pay for in the Sony. The larger transports in the Denon DCD-2700 and Harman Kardon HD7725, with their bigger laser assemblies and metal subframes, look as if they might be a little more reliable than the Marantz's Philips transport, but they use a lot of plastic in places most likely to go wrong (including the plastic gear assemblies that move the laser), so on balance I would guess the reliability to be similar to that of the Philips unit.

Recall that Marantz is a Philips subsidiary. Now here is a surprise: Marantz does not use a Philips DAC; instead they us an NPC SM5872BS. Readers may recall that I found the Philips low-end DACs to be substandard (Issue No. 15); apparently the Marantz engineers agree with me. *[See the explanation by David Birch-Jones of Marantz in this issue's "Box 978." Marantz engineering is basically all-Japanese and quite separate from Philips engineering, although many Philips parts are used.—Ed.]* The SM5872BS is similar in many respects to other MASH-type units; the oversampling rate is 32x, and the resolution of the DAC is 18 bits. The principal difference is that it has a 4th-order single-stage noise shaper with an 11-level multibit output. The multibit output goes into a pulse-width modulator as in any other MASH-like part. Note that the MASH acronym describes the architecture of the digital delta-sigma modulator (MultistAge noise SHaping). In a MASH system, multiple lower-order modulators are interconnected to form a higher-order modulator. This improves the stability of the modulator over implementations in a single higher-order loop. The direct design of higher-order loops is now much better understood than it was when MASH chips were first introduced, and it appears that NPC has successfully overcome the design problems associated with a single-stage 4th-order modulator. I am not going to get involved with the tradeoffs in the two approaches here, but the key point is that it is not correct to call the SM5872BS a MASH device because it has a single-stage architecture.

The key difference between this NPC unit and the Philips Bitstream device is that in the NPC the digital noise shaper generates a multibit code, which goes to a PWM DAC, which then gets filtered by off-chip passive RC and then active filters. (The Sony CXD2562 "pulse" DAC does the same thing.) The Philips DAC generates a higher-speed single-bit code which goes to an on-chip switched-capacitor filter. The result is that the Philips part has more noise, more distortion, and (as will be explained in a future article) the potential to generate idle channel tones. The gain linearity performance of the CD-63 is very good, the error increasing from -0.2 dB for a -80 dB input signal to -0.7 dB for a -90 dB signal and then remaining around -1 dB down to the -110 dB level. THD + N with a -24 dB input signal (to remove the effects of the analog-stage distortion) was 4 dB away from theoretical levels at all frequencies. This could be the result of noise at the output of the DAC or in the analog stage, or it could be the result of clock jitter (see the Bob Adams article in the last issue). As is typical for DACs that use oversampling, the time-domain monotonicity test was significantly nosier-looking than what we get with a good multibit DAC, such as the one in the Denon DCD-2700.

The SM5872BS also has the interpolating 8x digital filter built into the same die as the DAC. (A digital filter is also built into the SAA7345 CD decoder chip of the CD-63, but this digital filter is not used.) Specs from the data sheet of the SM5872BS on the digital filter are the same as for the SM5840D filter used in the Harman Kardon HD7725 CD player—only 87 taps (other digital filters have 200 to 300 taps), 19-bit data path, and a very small 14-bit coefficient word length (other filters are in the range of 19 to 28 bits). Ripple is ±0.125dB (4 orders of magnitude worse than the other filters), and stopband attenuation is only -55 dB (3 orders of magnitude worse than other filters). The result of the filter's presence can be seen on the frequency response plots of the CD-63 in the form of small ripples in the range of 2 kHz to 15 kHz. On the $849 Harmon Kardon I found the use of such a simple filter unacceptable. On this $399 CD player it looks like a good compromise design solution. An IIR digital filter is also included on the SM5872BS for the de-emphasis filter. The de-emphasis is not very accurate; the error at 4 kHz is -0.7 dB, and not less than ±0.2 dB at any frequency. The SM5872BS also has a digital attenuator, which the CD-63 uses to vary the analog output as well as the headphone output between 0 and -30 dB.

The Philips low-end DACs supplied only a single-ended output which resulted in even more distortion. The NPC SM5872BS is balanced and requires an external differential-to-single-ended converter. The differential-to-single-ended conversion and the first active pole of filtering is performed by an NJM2114D dual op-amp. (I have no information on what is probably a 5532 clone.) The signal then goes to the other half of the NJM2114D. This time the op-amp is configured as an inverting second-order filter. The advantage over a Sallen-and-Key filter is that the input stage of the op-amp sees no common-mode signal. The disadvantage is that it has a much lower input impedance.

The filter is then followed by an 8-transistor discrete op-amp in a unity-gain configuration. The whole circuit is encapsulated in a metal can just like the megabuck stuff. The input stage is a JFET differential pair; the output stage is a self-biased JFET follower. The differential pair is current-source biased (Krell did not do this in their Studio!). The two JFETs are a special low-noise device mounted in one package. The op-amp is configured with one gain stage, called a folded cascode. This structure has fast settling characteristics and wide bandwidth. It has somewhat less gain than a two-gain-stage amplifier but it has more than enough for a buffer. Overall, this topology looks as good as the best. (Yes, I did say the whole thing sold for $399.) Now if they would just use the same topology in the two pervious stages—because the full-scale distortion performance is not so hot with the NJM2114D's in the signal path. The best distortion performance was -90 dB (8 dB off the theoretical levels). Above 2 kHz it started to rise, reaching -81 dB at 10 kHz. To be fair we must point out that this beats both the Denon DCD-2700 and Harman Kardon HD7725 (but the Sony CDP-X707ES's performance crushes the Marantz). The full-scale distortion of the CD-63 did not change at all when the output was loaded down with 600 ohms.

Output capacitors are back-to-back (nonpolar) 220 µF electrolytics. Muting is by a pi network of bipolar switches, not relays (you can't have everything for $399). Each channel is driven by a separate control line from the digital filter. It is possible in this arrangement to shut down one channel in a channel separation test and keep the other alive. You would get very good channel-separation test results this way (we measured a phony 133 dB at 16 kHz) but you pay for the extra components to pull this "Julian-should-be-very-happy" trick.

The Marantz CD-63 presents readers of *The Audio Critic* with an interesting dilemma. It performs well enough to ensure that the audible signal at its output terminal is indistinguishable from that of even the best-performing CD player. It is built well enough so that it should last several years under steady use. If you spend more, you can get a better-measuring unit, a more reliable unit, a unit with more sophisticated technology, a unit with better look-and-feel. The Sony CDP-X707ES is the paradigm of such a unit. Is such a unit worth the five times greater cost? You must answer that question for yourself. •

Large Loudspeaker System with Subwoofers

Snell Acoustics Type A

(last-minute mini preview by the Editor)

Snell Acoustics, Inc., 143 Essex Street, Haverhill, MA 01832. Voice: (508) 373-6114. Fax: (508) 373-6172. Type A Music' Reference System, $18,999.00. Tested samples on load from manufacturer.

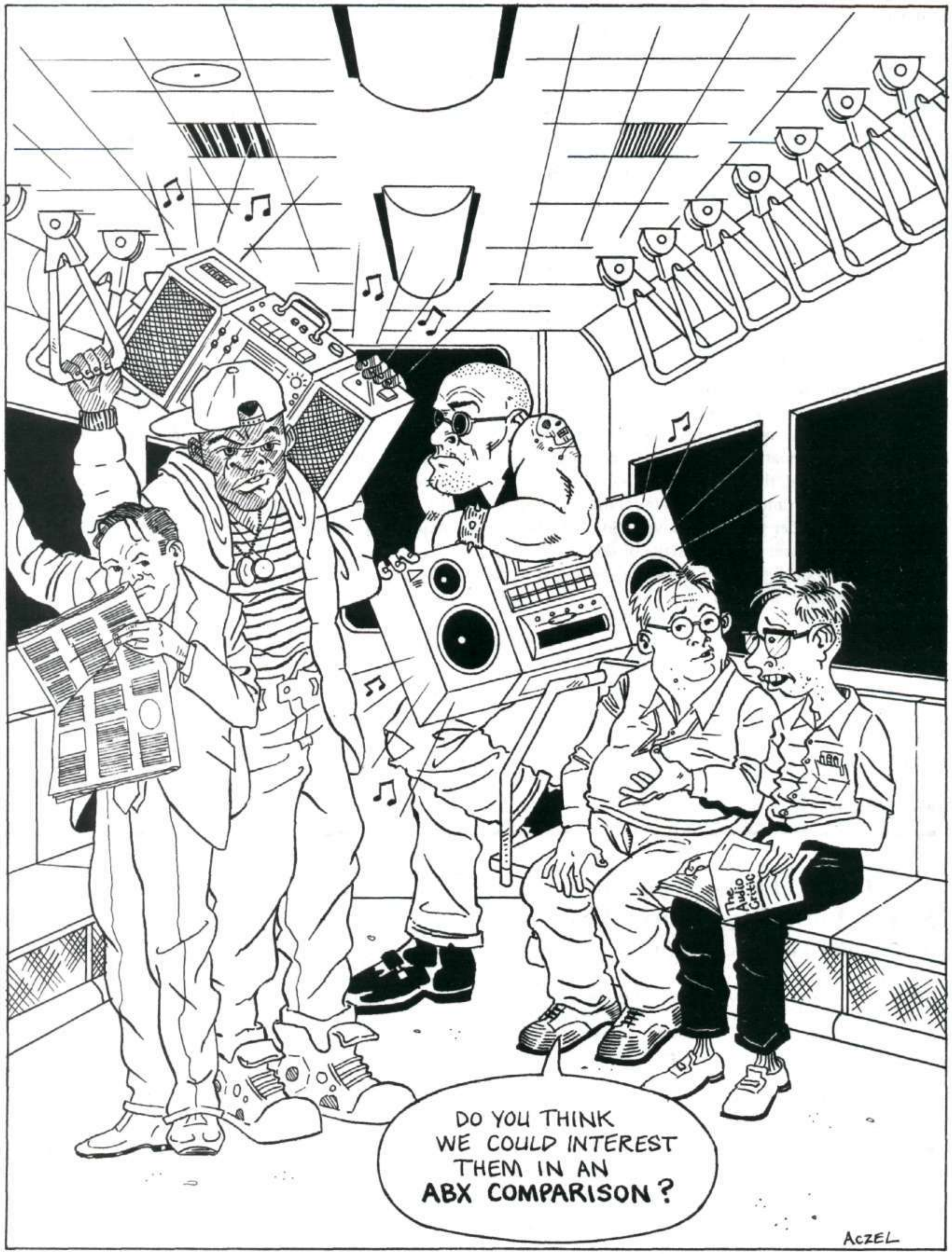
This new flagship of Snell Acoustics was definitely not scheduled to be reviewed in this issue; it came in when all the other reviews were already set in type. If we hadn't been late (yes, as usual), the issue would have been printed and delivered, with no chance for these comments.

Under the circumstances, I am still able to squeeze in the following statement:

Of all the loudspeaker systems I have listened to in my studio/laboratory, at my leisure, this is the one I like best so far. That doesn't mean it's perfect—and I haven't run any tests on it yet—but it appears to be more complete from the bottommost to the topmost frequencies, more neutral, and more seamlessly integrated than any other I am aware of. Of course, you could say that at nineteen thousand dollars it had better be good, but in high-end audio a lot of expensive equipment turns out to be disappointing. This isn't one of those. (Then I must hasten to add: unless you expect a miracle. It's still just a loudspeaker.)

More, lots more, in Issue No. 23.

In Your Ear


The Audio Critic
DO YOU THINK WE COULD INTEREST THEM IN AN ABX COMPARISON?
ACZEL

Large-Screen TV for Home Theater: Is a 40" Direct-View Tube Big Enough?

By Peter Aczel
Editor and Publisher

The top-of-the-line Mitsubishi direct-view set is good enough to make the choice between formats more difficult.

Our policy is to have at least one high-end video review in each issue for the home-theater contingent, but we ran out of space in the last issue and ended up with nothing on the subject. We don't want that to happen again because home theater has become the fastest-growing sector of home-entertainment electronics, whereas perfectionist two-channel stereo is on the wane everywhere except in Asia. At this point, we who are interested in the latest and greatest power amplifier circuit and such represent the nerdy rear guard of audio. We have no intention to stray significantly from those reactionary interests but we must keep up with the times and pay some attention to video, lest we should be perceived as having the priorities of the Yokohama Triode Society.

40" Direct-View Color TV
Mitsubishi CS-40601

Mitsubishi Electronics America, Inc., 5665/5757 Plaza Drive, P.O. Box 6007, Cypress, CA 90630-0007. Voice: (714) 220-2500. Fax: (714) 229-3854. CS-40601 color TV set, $4999.00 (best street price, $3050.00). Tested sample on loan from manufacturer.

To be perfectly accurate, the model I tested was the CS-40FX1, immediate predecessor of the more recent CS-40601. The two models are identical, the only difference being the inclusion of a closed-caption decoder in the CS-40601, as required by the FCC since July 1993. As there are no performance-related considerations in closed captioning, I am taking the very small liberty of running this review under the newer model number, just to be more up-to-date. (As we go to press, the very latest advertised model is the CS-40503; it appears to be pretty much the identical set. TV model designations tend to change several times a year.)

The 40" direct-view tube used in this Mitsubishi design is the largest obtainable in a commercially distributed TV set, at least to my knowledge. When you realize that rear-projection TVs also come in screen sizes from 40" up (or at least they used to, before 45" became the more common minimum), you begin to see just how huge this tube is. It goes without saying that the picture is sharper and brighter than with typical rear-projection screens—no contest. A couple of issues ago I wrote that *"big* is more important to me than *sharp"* but here the question is no longer big vs. sharp but big-enough-and-sharp vs. bigger-but-less-sharp. I still like 55" screens, but a 40" picture isn't exactly small, and the advantages of the direct-view tube begin to offset the size difference.

The sharpness advantage could be deemed illusory, since the number of scanning lines is independent of the screen size, but the closer spacing of the lines on the smaller screen always creates the impression of higher definition, so that the 40" picture appears sharper than the 55" picture. The brightness advantage, however, is very real—and huge. I have never seen a rear-projection set that wasn't much more enjoyable to watch in a darkened room, whereas it hardly makes a difference with the 40" direct-view Mitsubishi. The net result of all that is the perception of a "better" picture—better but smaller. If you need to see a life-size linebacker blitzing the quarterback, then you'll opt for the rear-projection screen. I'm not so sure. I'm waiting for a 55" direct-view tube (fat chance).

One disadvantage of the big Mitsubishi is that it's extremely heavy and, unlike typical rear-projections sets, doesn't come in the kind of cabinet that can be equipped with casters. Three of us out-of-shape weaklings were barely able to take it out of its factory carton and place it on the matching MB-40FX base. (That's not included in the price, but you don't absolutely need it; you can use a suitable table or anything else.) Once installed, the set is very easy and convenient to use.

As usual, I ran all the standard tests available on the Joe Kane laser videodisc *A Video Standard* (Reference Recordings, 1989). Black level retention was not as good as on a perfectly calibrated studio monitor, meaning that the ability to hold black at black, regardless of the picture content, was satisfactory but not perfect. Contrast could be easily turned up to the point where the peak linear capability of the set was exceeded, but with a little bit of tickling a reasonably linear and highly pleasing contrast level was obtainable. Color performance via the S-video input was excellent; the best settings of Color and Tint according to the test disc resulted in rich, brilliant, highly pleasing and accurate colors on the screen. I would rate the 40" Mitsubishi higher on color quality than any rear-projection set I've seen. Nearly all the default settings needed to be trimmed, however, for laserdisc perfection. On the other hand, the default settings represented a better compromise on middling TV broadcasts, so I can't really fault Mitsubishi on this.

Geometry was equally excellent; I observed no distortion of circles, checkerboard patterns, etc. I was also pleased to see that the picture wasn't overscanned. Convergence appeared to be right on the money. As for horizontal resolution, my test setup (as I've stated in the past) is good only up to about 400 or at best 425 lines, and that was no problem for the Mitsubishi. (The theoretical best for NTSC broadcasts is 336 lines.) Happily, I saw no fictitious ultrahigh-resolution specs for this set.

The two little outboard speakers provided with the set must be affixed by the user to the hinges on either side of the screen and wired to the back of the chassis. The hinges permit only a limited range of aiming; the speaker design is minimal, consisting of a tiny elliptical woofer and cone tweeter in a vented plastic pod. The stereo sound is surprisingly clean through these units when driven by the set's very low-powered internal amplifiers, but that's not what you want for home theater. The dinky little two-way speakers built into the MB-40FX base (they have *floor-level* dome tweeters!) are only a little more advanced; I wasn't particularly interested in using them. Instead, I hooked up the set to a complete Marantz AV surround system terminating in the Atlantic Technology System 250 six-piece speaker array. That way the sound was bigger and cleaner than in your average neighborhood movie theater, but I'm saving those reviews for the next issue (except for the Marantz MA500 power amp, which is featured in this issue). The 40" picture proved to be big enough not be overwhelmed by the formidable surround sound. There are no surround-sound processor or control facilities built into the set itself; it is assumed that the home-theater user will have the needed AV equipment. There is, however, a separate output for a powered mono subwoofer.

The control features and user conveniences of the Mitsubishi are also outstanding. The microprocessor-controlled audio/video adjustments are about as complete as I've seen on any set. The remote control, although far from an ergonomic masterpiece in its layout, is extremely versatile and effective. The menus that can be brought up on the screen for permanent settings (clock, timer, naming channels, locking out channels, etc.—the list is endless) are very clear and easy to use. So is the owner's manual, by the way. A single button will display on the screen the status of most of the current settings. The PIP (picture-in-picture) feature would be even in more attractive if the set had a second tuner (so you could check out a second program without an outside source, such as a VCR). All in all, the set can be said to be nicely loaded, as Chrysler would put it.

Cosmetically the Mitsubishi is in the high-tech black-plastic-bubble idiom, which is not exactly an interior decorator's delight, but the front of the set is all tube and virtually no plastic, so it will fit into a custom cabinet quite readily. Electrostatically attracted dust is especially visible on the black plastic surfaces. Bottom line: this is the only game in town when it comes to a direct-view TV of *serious* home-theater size—take or leave it. I'll take it.

Tech. Ed. of *The Audio Critic* to Chair a Tutorial Session at a Major Professional Conference

Dr. David A. Rich, Contributing Technical Editor of ***The Audio Critic,*** *has been invited to be chairman of the advanced tutorial session on audio at the coming* ***DSP^x*** *'95 conference (May 15-18, 1995, San Jose Convention Center, San Jose, CA). DSP^x is an exhibition and engineering conference for DSP and other electronic disciplines. David Rich's session (May 17, 1:00 P.M.) will last two hours and address the question: "Are there any design considerations in audio that go beyond standard measurements?" Joining him as speakers on the panel will be Richard Marsh of C.U.T.L., Inc., Sean Olive of Harmon International, Scott Wurcer of Analog Devices, Inc., and possibly one other technologist.*

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Obstructionism

By Tom Nousaine

Editor's Note: *Readers will possibly wonder whether this think piece by Tom Nousaine reflects the philosophy of the Editor and of this journal. The answer is: not entirely, but some of Tom's arguments are strong, and we agree with him on more points than not. That's sufficient reason to print him.*

* * *

I attended a forum on audio standards for surround sound and HDTV at the Late, Great, and Last Summer CES in Chicago. The panel included editors from a high-end audio and a high-end video magazine, and representatives from manufacturers pioneering new data-reduced multichannel formats. The audience seemed to be mostly audio manufacturers and journalists. The main question was: "Will developing data reduction and compression standards for surround and HDTV formats compromise sound quality?"

It is a legitimate question. One which brings an in-your-throat response of "It better not!" or we'll be stuck with less than optimal reproduction for years into the future ("Look at FM radio!" cry the protesters), sacrificing musical enjoyment for the shrinking audiophile population, and unknowingly deny the joys of high-fidelity reproduction to millions of unsuspecting Americans. During the course of the discourse, the High-End Community pushed for an opportunity to review the formats, inspecting for "transparency" before standards are set.

The martyr in all of us wants to stand and applaud. Yeah, let "Us" march on Washington, demanding Justice. But wait, Weaselthink demands analysis. What are the real implications of weak standards? How will an opaque standard actually affect an outcome? A quick review will reveal that chickenshit obstructionism could only get in the way of our quest for good sound.

Let's go back a short 15 years. In 1980 the Compact Disc was still an idea. A survey of hi-fi consumers would have shown that the Compact Disc wasn't needed. We were all happy with the quality of our best direct-cut LPs and 30-ips open-reel tapes. Many of us would have made the "you can't go from steak to hamburger and back" transparency argument.

How wrong those responses would have been. Analog techniques are so fundamentally different (and inferior) to digital, we just had no real way of evaluating the real benefits in advance. Instead, we reactively clung to the positives of the Status Quo, which, in fact, we knew to be manifold as we watched it become remarkably good with improvements over the years. What possible New Deal could be substantially better?

In addition to an innate inability to envision the benefits of change, we also harbor jaundiced views of present standards. Standards are simply consistent formats for interfacing machines and controlling their performance; they lower costs and make the machines easier to use in systems with complementary machines. For example, your Bryston works much better because it has convenient hardware and electrical interfaces that allow easy connection to other audio machines.

To complicate matters, standards are often developed before a technology is fully mature. Check out the RCA jack/plug. Most of us fear-and-loath them, but at the time—and even as ridiculous as they seem by today's rules—they were a key element of progress. Little development money would have been put into an industry that had no rational way of assuring new products could be easily used by the market.

Fundamentally, the RCA approached perfection in many ways. First, it did not materially impede delivered performance. RCAs sound like any other decent metal-to-metal connector at audio frequencies. Importantly, they are among the easiest to use of any type. They are also acceptably reliable in field use. They are inexpensive and therefore universally available. A perfect standard. Perfect at the time. Perfect in use. Perfect over time. In retrospect you could not have found a better way.

Further, our concept of "formats and standards" has been shaped by the hardware-based technology developments of the past. American society developed as we found ways of making better machines and better machines to make them. The automobile, the telephone, modern kitchen appliances, hand tools, amplifiers, LPs, even CDs are all good examples of progress through better machines.

Hardware-based standards tend to require massive financial commitments (changeouts of machines) when revised, a la Compact Disc. Although our entertainment values are performance-driven, they are value-based, and we are psychologically averse to throwing out perfectly good, hard-earned hardware. How many of us hung on to our open-reel machines years after we no longer used them? I ditched my LPs pretty quickly but my turntable lingered until a couple of years ago.

On the other hand, we also regularly change out existing equipment. How many phono cartridges have you swapped out in the name of better sound? How many amplifiers? Preamplifiers? Turntables? We change hardware all the time.

In fact, we crave acquisition of new present-technology equipment even when it represents, at best, a marginal improvement in performance. Yet, we bristle at the suggestion that new technology will force changeout of our current gear. Even as we fabricate reasons to buy/trade new present-technology hardware... outboard DACs anyone??

But, no one purchased their CD player or DAT at gunpoint. And how many can't find replacement cartridges? Or can't buy open-reel tape? Zero. None. On the other hand, how many gleefully found all kinds of previously thought-to-be-out-of-print LPs that showed up when record manufacturers started cleaning out warehouses?

Fact is, nobody gets shoved out when things change. No one is forced to buy new. It doesn't happen. On the other hand, current technology suddenly gets light-years better and much less expensive when new technology shows up. Check out linear-tracking tone arms in the mid-'80s. Everything, even the junk, gets better when technology improves. Our subliminal fears are totally unfounded.

Ultimately there are only two reasons for equipment turnover, anyway—performance gains and cost benefits. The movement to digital was, by any rational standard, performance-based, as are the new surround formats. On the whole, recordings today are frighteningly better than the best available in 1980. Sure, there are bad recordings. There were bad ones in the old days too, and we all complained about it passionately.

But now we have spectacular S/N, full bandwidth (2 Hz to 22 kHz), flat response, 74-minute+ play time, convenient no-touch access and playback, smaller size, no wear, and significantly improved sound. Much, much better performance.

CDs are also less expensive. True, the "manufacturer's suggested retail price" for new releases hasn't changed in 12 years (given the high inflation in the early '80s, that's a price reduction), but the average out-the-door price has fallen significantly. Players are now practically free. Compact Disc pretty much sums up the impact of technology change—higher performance and lower cost.

Better performance and lower cost. Why do we recall the pre-CD era as the good old days? Well, perhaps they were in some respects, but mostly it's because our memory is deficient. We regularly recast the past in a positive light while comparing it to the present with all warts in full itch.

It's the grass-is-greener syndrome, where time heals all wounds and every story gets better with each retelling. Herculean feats become even more epic; heroes get more heroic; cars get faster; fish get bigger; girls get better-looking; and your system seems better sounding in retrospect then it ever was in reality.

For example, we all remember our cars from the '60s as being incredibly fast. Much faster than today's smog-choked pretenders. Yet, when a car-buff magazine recently published a comparison of the fastest Z28 Camaros from the '60s and '70s with the new 1993 version, the new car turned out better in every regard, especially speed. The '93 is a full second faster to 60 mph than the fastest Camaro of yesterday. And, of course, it will go 154, which is 20 mph faster than the fastest vintage Z. It stops, accelerates, corners, and rides better than the very best of yesteryear. Yet, our memories recall otherwise.

Okay, so we shouldn't bemoan the passing of old ways and ignore newer and better ways. But, that still doesn't answer the problem of getting trapped with suboptimal format standards. What happens when a suboptimal standard gets used?

First, life goes on. The format adapts, or dies. Check out the compact cassette, which started life as a better dictaphone and ultimately turned into a decent, if imperfect, recording format that blossomed into a huge market even while saddled with format-related deficiencies.

I never owned a cassette player until 1988.1 still have my first one. It isn't hi-fi. I didn't really want one. Affected by the suboptimal format? Sure, lots of manufacturers made a hell of a lot of money selling cassette machines and dumped it into the development of CD machines and new recording artists. Good deal for me.

I never owned an El Cassette, either. Did I suffer because someone tried to develop an unneeded or substandard format? No way; I was never forced to buy one.

How about DAT? Largely regarded as a consumer flop, DAT has quietly revolutionized on-location, semipro, and amateur recording with an order of magnitude improvement in quality and cost. What was in it for me? More and better recordings. Did anyone force-feed the DAT on an unsuspecting public?

MD and DCC? Neither seems to have any real consumer application. If they fail, what have we lost? Have you been forced to buy one? Have they degraded anybody's stereo enjoyment? Have they caused any reduction in recording quality? Or made other equipment unavailable?

Would it have made sense to "delay" their introduction because they might not be transparent? Of course not. Sony and Philips might lose their butts on MD and DCC, but we bear no risk. And in the meantime Philips has demonstrated that a data-reduced format can be completely transparent.

FM Stereo? If it were to be invented today we would set higher performance standards. But, when it was developing, manufacturers strained to reach the performance limits. Now we have digital radio. Sure it will be better. But it was impossible when FM was a child. Should we have blocked the introduction of FM Stereo until digital radio was here because the standard was "suboptimal"? Would that have been the right thing to do? *[Whoa! The debate at the time focused on the pretty-damn-optimal Crosby system of multiplexing versus the preempting of the subcarrier for Muzak and similar commercial purposes. The latter approach prevailed. It was a political, not a technological, compromise.—Ed.]*

There is less concern today than ever because things change even faster now. Machines are becoming "processors." Format standards of the future will be software-based. You won't have to buy a new machine. Just an upgrade. And it will decode all your old material or allow you to run both versions.

Manufacturers won't need to buy all new machines to make new machines. Consumers won't have to scuttle their old machines, either. But, as a practical matter, they will anyway. How many folks use their original CD player? VCR? Laser disc player? Ten short years ago I wrote articles on a Kaypro CP-M machine with 64 KB RAM. How many computers have you owned or used in the past 10 years?

Hardware changeout is part of "higher performance." No, make that a natural part of life. Yet, we tend to have sweet, wet psychological dreams about the next round of present-technology products and psychotic nightmares about Freddy Krueger and Jack Nicholson secretly developing all subsequent generations.

Back to Chicago. I heard a bunch of progress-paranoid obstructionists, most of whom seemed genuinely well-meaning, wearing Transparency and Consumer Protection T-shirts, encouraging us to write manufacturers, urging that the High-End Community be given an "opportunity" to assess the transparency of new formats before new standards are adopted.

It was reminiscent of the knee-jerk reaction to "Perfect Sound Forever." PSF, incidentally, turned out to be pretty close to true on a practical basis. Imagine if we had let the High-End Community pass judgement on CD. 16 bits and 44.1 kHz would never have been "good enough" and we would still be vainly trying to squeeze 16 Hz onto a two-channel plastic groove.

The High End won't concede that two amplifiers have identical performance even when the goldenest of ears can't tell them apart with the nameplates off. Now they want us to urge others to let *them* decide when a data conservation scheme is transparent? I don't think so.

Sure, we may go through a couple generations of soft/hardware incarnations before we have Perfect Surround Forever. So what? Blocking the market door, even wearing a Purity and Transparency T-shirt, will only serve to hinder R & D and raise costs. I want manufacturers to spend their time making money in music and video. We should encourage even half-developed standards which will help companies pull "early" money out of the market, which in turn will encourage more development.

I want High-Definition Video and Higher-Definition Audio. It should foster experimentation, encourage new standards, and help new products get to the market. The long-standing performance improvement record of the electronics and computer industries, in the face of half-baked formats like Mac and DOS, is ample evidence that progress cannot, and will not, be hampered by emerging interface, protocol, and operating-standard inadequacies.

The biggest mistake would be to unwittingly join a groundswell of well-meaning public sentiment with its head up its ass. Inhibiting new standards, even imperfect ones, will curtail the quest for better sound. If new format standards are truly inadequate, they will die of their own accord. If they are substandard but mass-market acceptable, they will encourage new development which will, in turn, bring new, better sound and lower costs. We win in either case.

The High End really fears a reprise of the Compact Disc—fundamentally superior performance at substantially lower cost. What if the new digital surround sound blows away existing two-channel formats, including CD? What a shock to find yourself *two* generations behind in the quest for improved sound quality! Blocking the doorway to progress is the ultimate form of pissing against the wind. Even if you are wearing your Purely Transparent PC T-shirt.

Sunfire *(continued from page 31)*

ously when operated at worst-case levels. (Class AB amplifiers have maximum dissipation at about 0.41 of full power into a resistive load. Different dissipation peaks occur for different reactive loads.) The Sunfire amp can do this because it is much more efficient.

You may ask why nobody else has done this before now. Well, the degree of difficulty in designing an amplifier of this complexity requires a truly talented and creative engineer. One thousand Wonder-Cap-substituting nondegreed designers working for one thousand years could not possibly design this thing. Of course, this is not the only approach to the design of an efficient power amplifier but it is unique in not requiring modifications to the active amplifier electronics to function and hence imposes no performance degradation on the design. Indeed, a tracking power rail will hold the V_{CE} of the output devices constant and reduce open-loop distortion in the amplifier's output stage.

The amplifier we briefly tested was a very early production unit. Construction was excellent, with good-quality parts on double-sided boards. A clamshell chassis design gives the unit good structural rigidity and the look of an even more expensive amplifier. Initial measurements showed that the amplifier is what it claims to be—the most powerful amplifier we have tested, although the Parasound HCA-2200II came very close. We did not test the Parasound under worst-case continuous-drive conditions, but based on its construction I would expect it would go into thermal shutdown under conditions that would leave the Sunfire still operational. However, as explained below, it would be possible to reverse this situation because of a design simplification in the production Sunfire.

Some switching noise was noted at the output, and the protection circuits need some more design work to prevent premature activation. In the version we examined the switching power supplies were shared by both channels. Consequently the maximum supply voltage required by the "hotter" channel must be produced by the tracking supplies. It is easy to imagine a set of test signals that would cause the amplifier to have very poor efficiency, since the right channel might need a high positive supply rail and the left a high negative supply rail. Under such a condition the maximum available supply voltage could be across the output devices, not just the 14 V. Good efficiency is maintained only if the left and right channels are correlated. The distortion-reducing property of having the V_{CE} of the output devices held constant is also lost with totally uncorrelated left and right inputs. We did not test this on the early production sample but will look into it when the unit undergoes full testing.

—David Rich

Hip Boots

Wading through the Mire of Misinformation in the Audio Press

Editor's Note: *Just because there are no Harley Howlers discussed below, don't for a moment assume that* Stereophile's *Consulting Technical Editor has stopped putting his jittery foot into his digito-dilettantish mouth. Hardly a week goes by that some degreed technologist doesn't complain to me about the outrageous Harleyfication of digital theory. But how many times can we repeat the same depressing message? We must turn our attention to other sources of audio misinformation.*

Et tu Forbes?

Forbes magazine is supposed to have a pretty good research staff. If they did a piece on, let us say, the future prospects of commuter airlines, they would go to the world's top authorities on the subject and obtain a variety of opinions and forecasts. Audio, however, is clearly not worthy of such conscientious fact-finding. Anyone with strong opinions is an authority, or so it would appear from Toddi Gutner's article "Blasts from the Past" (7/4/94).

The article is about the resurgence of vacuum-tube amplifiers in the audio market and starts with a quotation from "Tom Havens, a Manhattan attorney and audiophile" to the effect that a tube amplifier "gives you the feeling that you're in the same room with the performer." Tom Havens? That's the authority available to *Forbes?* Can't they call an E.E. professor or a psychoacoustics Ph.D. at one of the major universities? Somebody like Dick Greiner would have been very happy to set them straight. Toddi Gutner then adds, editorially, that "even a casual listener can easily discern" the difference a tube amplifier makes because "music sounds warmer, more natural and realistic."

A qualified source would have explained that the difference, if at all audible, is not due to tube magic but to higher second-harmonic distortion and higher output impedance interacting with the speaker load, both of which characteristics can be easily and cheaply duplicated in any solid-state amplifier. (Ask Bob Carver, who has done it often.)

The other "authority" quoted in the article is William Wright, co-owner of Cary Audio Design, one of the companies that promote the single-ended triode idiocy. Toddi Gutner would probably consult Dennis Hopper on the medicinal benefits of cocaine. *Forbes* editors, repent!

Tim Smart is at it again in *Business Week.*

This is the same Tim Smart I castigated in Issue No. 16 for "untutored technobabble" and relying on tweako sources of information instead of advisors with scientific credentials. I even wrote a separate letter to *Business Week* and got a mealymouthed, all-purpose, printed postcard for my pains. Well, Tim is still their man for high-end audio, as witnessed by the "Personal Business" section (10/10/94), "Electronics" subdivision, under his byline. What do you think he writes about? Yes, indeed, tubes! He is a *little* more careful—could it have been my column or my letter?—but not enough. For example: "What makers of tube equipment are seeking is a more natural sound. Audio engineers speculate that while all amplifiers distort to some degree, tubes do it in a manner more gradual—hence less offensive to the ear—than do transistors."

That's pure garbage. Audio engineers do no such speculating; ignorant tweaks do. Any audio engineer with a B.S.E.E. or better knows that a solid-state circuit can be designed to distort just as gradually as any tube circuit, if that's what the goal is (in most cases it isn't or shouldn't be). Furthermore, a properly designed solid-state amplifier can have as little as 0.002% distortion, which is not "some degree" but at least two orders of magnitude below audibility, hence in no way "offensive to the ear." Tim Smart also brings up vacuum-tube guitar amplifiers but fails to grasp that they are popular exactly because of their funky colorations/distortions near the clipping point—they add a little something to the music. An accurate amplifier, designed for reproduction only, is not supposed to do that.

And who are Tim Smart's quoted "authorities"? The marketing man of Audio Research. The president of

Valve Amplification Company. Sam Tellig/Gillet of *Stereophile.* Tweaks and suppliers to tweaks. When, oh when, will a reputable business magazine quote Edward Cherry or Bob Cordell on the subject of amplifiers? Or Stanley Lipshitz or Dick Greiner or Floyd Toole or Mark Davis on any number of other audio topics? Are their university degrees and professional credentials standing in the way? Maybe those highly paid editors at the business magazines don't know how to pick up the telephone and find a genuine authority on a subject they don't have to deal with every day. How sad.

Which part of *The Absolute Sound* d'ya read?

On page 7 of *The Absolute Sound's* annual *Guide to High End Audio Components,* '93/'94 edition, Harry Pearson writes:

> Now, we who review for this publication are all too aware that some true believers in search of audio gospel will robotically go out and buy whatever HP et al. find delectable. I, for one, would just as soon have this sort read *Stereophile,* which seems anxious for this kind of "authority" over others. This magazine is for people who think for themselves. I detest the notion that I am regarded, by some, as an "expert" (none exist), and "authority" (none exist), a "golden ear" when I have never, repeat never, said or suggested I wanted to be or was any of these things. You'd be surprised, however, how many of the mentally impaired describe us as "self-proclaimed experts or gurus."

Well, I must be mentally impaired because I keep looking at the front cover of the *Guide* and I see six words in big letters that could never, repeat never, have been printed there. In a 24-point, bold, shadow-style, Gothic typeface it says (or so it appears to my impaired mind):

The Authority on High End Components

This is an audio journalist/editor we are supposed to take seriously? And if you ever took him even half seriously, no matter who you are and no matter how long ago it was, don't you feel a little sheepish about it? The man is an irresponsible blabbermouth and an embarrassment to the audio community.

Ken Kessler in *Hi-Fi News & Record Review*

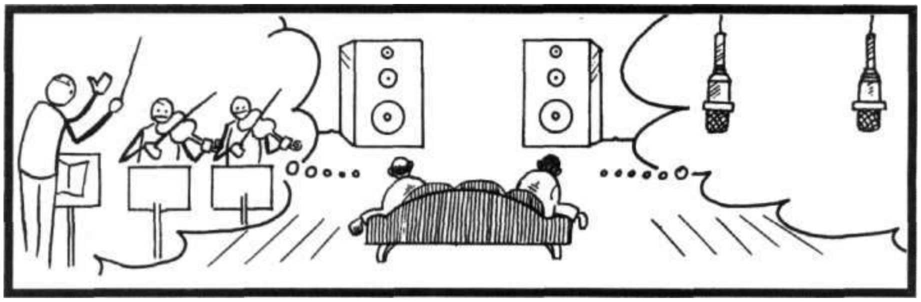
Who is Ken Kessler? He is U.K.-based; he is a tweako audio journalist (very *engage);* he writes well; his main hangouts have been *Hi-Fi News & Record Review, Stereophile* (recently dropped from the masthead), and *Audio* (recently added to the masthead). *HFN/RR* and *Audio* are currently the English and American representatives, respectively, of the editorial philosophy that science and antiscience deserve equal time. (Is that Politically Correct or simply muddleheaded? Don't ask me.)

What caught my eye, long after publication, was a letter to the Editor by Ken Kessler in the May 1994 issue of *HFN/RR.* He responds to a Mr. Burmajster, who had apparently disparaged subjective audio reviewing and asserted the greater validity of measurements. Ken Kessler writes: *"the measurements of hi-fi components have little if any bearing on the sounds they make"* (italics his). He adds that "unless we're talking about ludicrous anomalies, like a speaker which has a 20 dB dip at 5 kHz or an amp with 10% distortion, the hundredths-of-a-percentile measurements foisted on us (initially by the Japanese) do nothing to tell us whether or not a product reproduces sound convincingly and realistically." Well, Ken, how about the ±1 dB or so frequency-response tailoring introduced by an amplifier output impedance of 1 ohm or so (see our Issue No. 16, page 55) on which an entire multi-megabuck "tube sound" commerce is based? How about an error of 0.3 dB in the de-emphasis curve of a CD player that can make certain pre-emphasized CDs sound too bright? And, in general, when you subjectively find that a hi-fi component sounds like this or like that, there exists no verifiable cause for the effect you perceive?

Further on, the letter challenges Mr. Burmajster as follows: "if he can look at the measurements (manufacturer's or self-measured) of five D/A converters and then match them to the converters in a blindfold listening session, I will never again write a subjective, measurement-free review." The hypocrisy of that challenge leaves me in a state of sputtering rage. Here is a subjective reviewer who will unhesitatingly declare that A has better soundstaging, or less grain, or more bloom than B but never, never feels the need to prove that he can actually tell A and B apart "in a blindfold listening session"—and he has the nerve to challenge someone else to have a perfect score not in an A/B but an A/B/C/D/E blind test! You know damn well, Ken, that neither one of you can tell any *two* D/A converters apart in blind listening. That's the whole point, man! Whether they have -97 dB or -85 dB distortion and noise, it's well below the threshold of hearing, but I want to know those numbers—and a whole slew of others— to find out which is more carefully engineered and the better value. Do you have a better way of evaluating them? Yes, I know, "by listening"—except that you're unable to hear the difference unless you're allowed to look at the nameplates! It's an unfunny farce.

There are certain basic realities in audio today, and it's time for all audio journalists to face them. Present-day electronic signal paths are sufficiently clean to have no distinguishable "sound" of their own. You have to measure them, analyze the circuits, look at the quality of parts and construction—that's how you evaluate them. That great admonition "Why don't you just listen?" works only with loudspeakers, earphones, and microphones today. That makes the nontechnical, noncircuit-reading, nonmeasuring reviewers who "just listen" something of an underclass, but what do they care? They can still get all the expensive toys they lust for on extended free loan from the component manufacturers. •

Recorded Music



***Editor's Note:** David Ranada had a respite this time while I was catching up on my CD backlog, as promised. (Not that I caught up totally—or ever will.) David will be back in the next issue, so that I can fade into the background again (where I belong as a music critic, no?). As long as I was at it, I threw in some very recent releases, so what we have here is a mishmash of the new and not-so-new.*

Editor's Grab Bag of CDs, New or Fairly Recent

By Peter Aczel
Editor and Publisher

The following two items deserved a little more emphasis than the capsule reviews, I thought. (Note that in all cases the year in parentheses after the CD number is the year of recording, not the year of release.)

Fritz Reiner in "Living Stereo"

All of these Reiner/Chicago recordings from 1954—59 have been released on various **RCA Victor** CDs before, but the Living Stereo remastering process has given them a new lease on life. It has to say "Living Stereo" on the label, not Red Seal or Gold Seal, to indicate this meticulously accurate recreation of what's on the original analog master tapes (most of them recorded by the great Lewis Layton).

***Richard Strauss:** Also sprach Zarathustra, Op. 30; Ein Heldenleben, Op. 40. 09026-61494-2 (1954).*

***Brahms:** Violin Concerto in D Major, Op. 77. **Tchaikovsky:** Violin Concerto in D Major, Op. 35. Jascha Heifetz, violin. 09026-61495-2 (1955 and 1957).*

***Bartók:** Concerto for Orchestra; Music for Strings, Percussion and Celesta; Hungarian Sketches. 09026-61504-2 (1955 and 1958).*

***"The Reiner Sound"** (shorter works by **Ravel, Liszt, Weber/Berlioz, Rachmaninoff).** Byron Janis, piano. 09026-61250-2 (1956,1957, 1959).*

***Mussorgsky:** Pictures at an Exhibition (orch. by Ravel); Night on Bald Mountain; "and Other Russian Showpieces." 09026-61958-2 (1957 and 1959).*

There are two utterly remarkable things about this series. One is the quality of the recorded sound, which is only a smidgen below the very best we have today. Except for the somewhat higher noise floor (tape hiss) and infrequent distortion (mike overload, tape saturation), these could be 1994 recordings *(good* 1994 recordings, that is). For their time, they were absolutely miraculous. Even more miraculous is the playing of the Chicago Symphony Orchestra. They are still good today but not *this* good! No orchestra today plays on this level of virtuosity and synchronicity. Maybe no orchestra ever did, except possibly the New York Philharmonic in the 1930s under Toscanini.

As for interpretation, I would say the Strauss and Bartók recordings have never been equaled; the others are as good as any; and Heifetz in the violin concertos is of course in a class by himself. These CDs represent the pinnacle of orchestral performance and also of audio restoration.

The Bayreuth/Barenboim *Ring*

In the last issue, I capsule-reviewed the live Bayreuth (1991 and 1992) recordings by **Teldec** of *Das Rheingold* and *Die Walküre.* Now the *Siegfried* (4509-94193-2) and *Die Götterdämmerung* (4509-94194-2) albums are also out, and one can view Barenboim's *Ring* in its totality. One Wagnerite's assessment:

The sound is just right; for the first time, the unique Bayreuth acoustic, so perfectly suited to this music, comes through with every detail in place, both orchestra and singers in crystal clear relief against each other. That alone is worth the price of admission. Barenboim, while neither a Toscanini nor a Furtwängler, is a superior Wagnerian in his own right; he is not mannered and is always in total control. The Bayreuth Festival Orchestra is simply magnificent. Only the singers are so-so, except for Anne Evans, an excellent Brünnhilde. John Tomlinson as Wotan is fair to good.

I'd give this *Ring* an A minus.

Athena Productions

This is a small label from Staten Island, New York, ostensibly trying to obtain wider distribution. They purchased the rights to the recording below from **Sonora Records.**

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"Music for Violin and Guitar." Arturo Delmoni, violin; David Burgess, guitar. SACC102 (1991).

Single mike, Blumlein pattern, no gain riding, custom electronics, tweako engineer (Bob Katz, see Issue No. 17, p. 45), high-end audiophile artists, cryogenically processed violin strings—get the picture? I was ready to have a bit of tweak-bashing fun with this, but the sound is truly gorgeous, featuring an utterly natural, sweet violin tone and somewhat reticent but beautifully delineated guitar twangs. The imaging is very stable. The music is mostly transcriptions of light classics (one contemporary work: David Leisner's violin/guitar sonata); the playing is stylish and highly respectable in technique. A demo CD, much as I hate to admit it.

Bainbridge

Here we are in Brad Miller country, and that means amplifier-clipping, loudspeaker-busting sound effects (although the man is also quite capable of making first-rate musical recordings). The Colossus digital processor and discrete multichannel master tapes are ingredients of the SPL fest in each instance.

•

"Sonic Booms 3." BCD 6289 (1992 and earlier).

Missiles, space shuttle launches, supersonic jets, and suchlike, alternating with peaceful surf sounds, etc. As good as this sort of thing gets. Forget it without a subwoofer, though.

"4449 Pinnacle! Daylight." BCD 6295, Discs 1 and 2 (1991-92).

Strictly for aficionados of classic steam locomotive sounds but the very best of the genre. No, I didn't listen to every one of the 27 tracks on two CDs, but I'm ready to acknowledge that Brad Miller is the master of this. Once again, a good subwoofer is mandatory. (And I hope you already own Beethoven's Ninth.)

Chesky Records

This label is one of the apostles of "natural" sound via minimalist recording techniques, an approach which in most cases yields superior results—but not always. (In jazz, Chesky's long suit, it works without fail.)

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"O Magnum Mysterium." Westminster Choir, Joseph Flummerfelt, conductor; Nancianne Parrella, organist. CD83 (1992).

A potpourri of choral music spanning five centuries (from Victoria and Byrd through Mozart, Verdi, Bruckner, Brahms, etc., to Stravinsky and Messiaen), very well sung by the Westminster group and chastely recorded by the aforementioned Bob Katz through tweako vacuum-tube electronics with audible hiss. Church acoustics, very good imaging.

CIM

The full name is The Cleveland Institute of Music, which is a famous conservatory, and what we have here isn't really a CD label but a one-off, in-house recording project.

•

***George Szell:** "Music by George Szell." The Cleveland Institute of Music Orchestra, Carl Topilow and Louis Lane, conductors; The Cavani String Quartet. CIM release #2152 (1992).*

Two orchestral works and a piano quintet by the great conductor, who was an eclectic but highly imaginative composer in his early youth. Brahms, Dvoralc, Mahler, Strauss—you can hear them all in these clever pieces composed between the ages of 14 and 24. The playing is uniformly excellent in all three works; the recording of the orchestra is pleasantly bright and punchy, not at all harsh, and extremely detailed and impactful (Judy Sherman was the producer).

Delos

This remains one of my favorite labels because the highly intelligent leadership of Amelia Haygood and the unique recording skills of John Eargle almost invariably translate into classy repertory and superb sound.

•

***Ludwig van Beethoven:** The Complete Quartets, Volumes I through VI. The Orford String Quartet: Andrew Dawes and Kenneth Perkins, violins; Terence Helmer, viola; Denis Brott, cello. DE 3031-36 (1984-86).*

This is a most unusual situation—recordings dating back 8 to 10 years, by a now defunct string quartet, recorded by a team (Marc Aubort and Joanna Nickrenz) no longer associated with Delos, and still being released one CD at a time (Volume VI only recently this year), two quartets per CD, with five quartets and **the** *Grosse Fuge* **yet** to come. But what a series! The Orford was for more than two decades the pride and joy of Canada, as good a string quartet as any. I was weaned on the Budapest and the Guarneri **but** I enjoy the Orford equally. Their playing is totally secure and highly nuanced; in the new Volume VI, for example, the monumental Opus 131 receives a marvelously lucid and musical performance. The Aubort/Nickrenz digital tapings are fully up to 1994 standards, as transparent as it gets. Even John Eargle has to be satisfied, although he would probably have recorded the instruments a little bit less close. Matter of taste. As for the delayed releases, don't ask me why.

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***David Diamond:** Vol. Ill (Symphony No. 1; Violin Concerto No. 2; The Enormous Room) and Vol. IV (Symphony No. 8; Suite from TOM; This Sacred Ground). Seattle Symphony and Chorale, Gerard Schwarz, conductor; Ilkka Talvi, violin (in the Concerto). DE 3119 (1991-92) and DE 3141 (1992-94).*

The recording of the complete works of the ageless David Diamond continues in Seattle under the supervision of the composer. (Beethoven never had that advantage.) Even the most recent of these works is more than thirty years old; originally they were not "modern" enough for the critics but in the post-modern era they come off as romantically inspired, timeless art, masterfully crafted with lots of orchestral color, and perfectly suited in sound to the high-fidelity audio medium. John Eargle's highly panoramic soundstage provides the ideal framework. You should listen to all four volumes issued so far and own at least one.

•

***Walter Piston:** The Incredible Flutist (Suite); Fantasy for English Horn, Harp & Strings; Suite for Orchestra; Concerto for String Quartet, Wind Instruments & Percussion; Psalm and Prayer of David. Juilliard String Quartet; Seattle Symphony & Chorale, Gerard Schwarz, conductor. DE 3126 (1991-92).*

Walter Piston's orchestral music is exceptionally "phonogenic"—that is to say, CD-genic—quite independently of its far from negligible musical qualities. He was also one of the insufficiently abstruse mid-20th-century eclectic composers now that much more appreciated for his then-reactionary comprehensibility. This is part of yet another Delos/Schwarz complete-orchestral-works project, the third of the Piston series—and these people appear to be capable of chewing all that they bite off! *The Incredible Flutist* is a delightful 1938 ballet, probably the most attractive music on this disc, but the other pieces are also highly listenable. The audio quality is positively awesome, one of John Eargle's most dazzling efforts.

•

***Sergey Prokofiev:** Piano Concerto No. 3 in C Major, Op. 26. **Aram Khachaturian:** Piano Concerto, 1936. Dickran Atamian, piano; Seattle Symphony, Gerard Schwarz, conductor. DE 3155 (1993).*

The Khachaturian concerto is a flashy, pop-appeal, highly Sovietized piece in my opinion; the Prokofiev concerto is an early-20th-century masterpiece. Atamian is a high-powered virtuoso, perhaps without that final touch of finesse but still very impressive. The curiosity here is the audio: Delos without John Eargle. Al Swanson recorded the 20-bit master; no proprietary technology is claimed for the 20-to-16-bit conversion, but the recording has a huge dynamic range, and the sound is still unmistakably Earglian, possibly a little closer and drier in the Khachaturian.

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***Richard Strauss:** Metamorphosen. **Arthur Honegger:** Symphony No. 2. **Anton Webern:** Langsamer Satz. Seattle Symphony Strings, Gerard Schwarz, conductor. DE3121 (1992-93).*

All of this is string music, very competently but not superbly played. The Strauss is a late-in-life masterpiece which should be in every collection (but maybe not in this interpretation). The John Eargle recording sounds rolled off on top; could the Wadia digital processor he used be the reason? (It's not his usual equipment.)

•

"Inaugural Recital: David Higgs premieres the C. B. Fisk Organ of the Meyerson Symphony Center, Dallas." DE 3148 (1993).

McDermott Hall at the Meyerson Symphony Center in Dallas is famed for its acoustics and has a state-of-the-art system of built-in chambers for tuning the reverberation time. John Eargle, for one, knows how to tune it, and the new Opus 100 organ of C. B. Fisk, Inc., is as good as it gets. Add to that an organist with the clean, elucidating touch of David Higgs and organ masterpieces by Bach, Liszt, Franck, et al.—the combination is awesome. Purists note: one pair of omnis did it all! I especially recommend this CD to audiophiles who are just starting to get interested in organ music.

Denon

Nippon Columbia continues to make technically superior recordings of important artists on the rising slope of their career, rather than safe household names "mailing in their performance." I always listen to new Denon releases with a sense of anticipation.

•

***Johannes Brahms:** Piano Sonata No. 3 in F Minor, Op. 5; Variations and Fugue on a Theme by Handel, Op. 24. Bruno-Leonardo Gelber, piano. CO-75959 (1992).*

The sonata, albeit quite famous, isn't my cup of tea (see Issue No. 17, pp. 58-59), but the Handel variations are wonderful music, beautifully crafted. Gelber is a world-class virtuoso and has great affinity for these pieces. He almost sells me on the sonata, too. The recording is close and extremely dynamic, with great in-your-room presence. It is one of Denon's new Mastersonic Series with 20-bit processing—very impressive.

•

***Johannes Brahms:** String Quartets, Op. 51 (No. 1 in C Minor, No. 2 in A Minor). Carmina Quartet: Matthias Enderle and Susanne Frank, violins; Wendy Champney, viola; Stephan Goerner, cello. CO-75756 (1993).*

Again, the peculiarly

strained, uptight Romanticism of these works represents the least attractive side of Brahms to me—but that's just my own lowbrow opinion. Besides, Brahms the master craftsman often peeks out from under the sugary glaze. Very slick, highly disciplined playing and beautifully detailed, first-row sound.

•

Claude Debussy: *String Quartet in G Minor, Op. 10.* ***Maurice Ravel:*** *String Quartet in F Major. Carmina Quartet: Matthias Enderle & Susanne Frank, violins; Wendy Champney, viola; Stephan Goerner, cello. CO-75164 (1992).*

Two masterpieces, masterfully played. I don't think you can get a better digital recording of this music. The Carmina players offer style, precision, nuance, beautiful tone—the whole bit. The recording is again close, detailed, extremely vivid. A great CD.

•

Joseph Haydn: *Symphony No. 94 in G Major ("The Surprise"); Symphony No. 22 in E-flat Major ("The Philosopher"); Symphony No. 45 in F-sharp Minor ("Farewell"). Orchestre de Chambre de Lausanne, Jesus Lopez-Cobos, conductor. CO-75660 (1992).*

This traversal of the Haydn symphonies with modern instruments continues to go well. Lopez-Cobos conducts strong, clear, idiomatic performances of these well-known works; the orchestra is very good but not great. The recorded sound is smooth as silk, possibly the main appeal of this CD to audiophiles.

•

Felix Mendelssohn: *Violin Concerto in E Minor, Op. 64.* ***Henri Vieuxtemps:*** *Violin Concerto No. 5 in A Minor, Op. 37. Chee-Yun, violin; London Philharmonic Orchestra, Jesus Lopez-Cobos, conductor. CO-78913 (1994).*

This very recently issued recording of the 24-year old Korean violinist shows considerable artistic maturation since I first heard her play as a teenager at a private dinner party, and even since her earlier Denon recordings (namely *Vocalise: Violin Showpieces*, CO-75118 [1992] and three Sonatas by Fauré, Debussy, and Saint-Saëns, CO-75625 [1993]). Her technique is all that could be asked for; her tone is gorgeous; and she is clearly beginning to assert a strong interpretive personality. I'm impressed. Lopez-Cobos and the London Philharmonic support her with considerable panache. What's more, this is another of the new Mastersonic 20-bit recordings, and the sound is fabulous. I can honestly say I've never heard a better-sounding concerto recording. There is a sense of completeness and dynamic ease conveyed by the recorded sound that I find quite unusual.

W. A. Mozart: *6 'Haydn' Quartets (G Major, K. 387; D Minor, K. 421 [417b]; E-flat Major, K. 428 [421b]; B-flat Major, K. 458 ["Hunt"]; A Major, K. 464; C Major, K. 465). Kuijken Quartet: Sigiswald Kuijken & François Fernandez, violins; Marleen Thiers, viola; Wieland Kuijken, cello. CO-75850/51/52 (1990-92).*

What a disappointment! These unquestionably excellent and scholarly musicians play very chastely and precisely on period instruments, which are tuned to A_4 = 430 Hz. Nothing wrong with that, but the playing is vibratoless and expressionless (just the opposite of the Carmina Quartet), and the tone is unpleasantly nasal and penetrating, reproduced to perfection by the English engineers. This is some of my favorite chamber music—indeed, some of *everybody's* favorite chamber music—but I find this version virtually unlistenable, complete and authoritative though it may be.

W. A. Mozart: *Concerto in A Major for Clarinet and Orchestra, K. 622.* ***Ferruccio Busoni:*** *Concertino for Clarinet and Small Orchestra (1919).* ***Aaron Copland:*** *Concerto for Clarinet and String Orchestra, with Harp and Piano (1948). Paul Meyer, clarinet; English Chamber Orchestra, David Zinman, conductor. CO-75289 (1992).*

The wonderful Mozart concerto receives a fluent, smooth, professional performance without much expressive nuance. The Busoni piece is uninteresting, but the Copland has a certain Stravinsky-ish flair to it; both are performed competently. The sound isn't Denon's absolute best, but even their second best is awfully good.

Camille Saint-Saëns: *Symphony No. 3 in C Minor, Op. 78 ("Organ"); Le Rouet d'Omphale, Op. 31; Phaeton, Op. 39; Danse Macabre, Op. 40. Michael Matthes, organ; Orchestre National de Lyon, Emmanuel Krivine, conductor. CO-75024 (1991).*

Krivine is highly musical and idiomatic in these works, but his "Organ" symphony is a bit tame for my taste; it should be weightier and more rip-roaring. The orchestra is good; the organist is excellent; the Denon recording is once again of demo quality.

Deutsche Grammophon

The new "4D Audio Recording" system actually seems to have helped the generally quite dreadful DGG sound, whether or not as a result of all the much-hyped massaging of the analog and digital electronics. The fact is that the 4D Schubert disc reviewed below sounds incomparably sweeter and sonically more believable than the non-4D Bruckner disc of the same orchestra and conductor, recorded by the same DGG team in the same church. So the leopard *can* change his spots, Jeremiah! (Well, maybe only a German leopard...)

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Anton Bruckner: *Symphony No. 7 in E Major. Staatskapelle Dresden, Giuseppe Sinopoli, conductor. 435 786-2 (1991).*

A rather uninspired, heavy-handed, heavy-footed performance, although the Dresden orchestra is a great one, as we all know. The violins scream into the microphone(s) in the traditional DGG manner whenever the music gets loud. I'm distinctly underwhelmed.

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Franz Schubert: *Symphony No. 8 in B Minor, D759 ("Unfinished"); Symphony No. 9 in C Major, D944 ("The Great"). Staatskapelle Dresden, Giuseppe Sinopoli, conductor. 437 689-2 (1992).*

Now this is more like it. These are much more loving, idiomatic, nuanced performances, with many musical felicities, although they won't displace any number of truly great recordings of the past. Still, they're good enough even for first exposure to these masterpieces. The 4D sound is smooth, spacious, authoritative, highly listenable. About time, DGG.

Dorian

Craig Dory, president and chief engineer of this extremely audio-conscious and audio-savvy company, took me to task for my perhaps too casual Editor's parenthesis (Issue No. 21, p. 66) suggesting that he experiments with more and more ambience out of boredom or restlessness. I think I had the right to the opinion that he sometimes goes overboard ambience-wise but I was out of line when I got personal regarding his motivations. He claims that all of his Dallas Symphony Orchestra recordings are microphoned exactly the same way. OK, but what about the McDermott Hall's tunable reverberation time?

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Ernest Chausson: *Symphony in B-flat Major, Op. 20.* ***Jacques Ibert:*** *Escales; Divertissement. Dallas Symphony Orchestra, Eduardo Mata, conductor. DOR-90181 (1993).*

Chausson's only symphony, a fine work in the Franckian idiom, receives a very solid performance here without providing major thrills (is Mata a kind of Señor Ormandy?). The Ibert pieces could be played with a little more sparkle, but the orchestra plays very well and the musical points are clearly made. This time the ambience is just right, and Dorian's 20-bit technology yields gorgeous sound.

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Manuel de Falla: *La Vida Breve. Marta Senn, mezzo-soprano; Fernando de la Mora, tenor; Cecilia Angell, mezzo-soprano; Simon Bolivar Symphony Orchestra of Venezuela with choruses, Eduardo Mata, conductor. DOR-90192 (1993).*

Mata with a different orchestra, recorded by Craig Dory in a different hall—that's an interesting situation. Unfortunately, the performance is not so interesting. The singers and the orchestra are good but not special, and so is the performance, which never really catches fire, although Falla's first opera is fiery enough. The L6pez-Cobos performance on Telarc (see below) is definitely more exciting and more authoritative. The 20-bit Dorian recording, however, is quite wonderful—slightly more refined than the also superb Telarc but still very dynamic, a little more closely recorded than the Dallas sessions, in a hall of the Universidad Central de Venezuela in Caracas. No excessive reverberation—how about that?

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Joseph Jongen: *Symphonic Concertante for Organ and Orchestra, Op. 81.* ***Camille Saint-Saëns;*** *Symphony No. 3 in C Minor, Op. 78 ("Organ"). Jean Guillou, organ; Dallas Symphony Orchestra, Eduardo Mata, conductor. DOR-90200 (1994).*

Blockbuster music. The incredible Fisk organ. McDermott Hall. The one and only Jean Guillou. A 20-bit Craig Dory recording. This CD promises a lot and delivers most of it. For a 1926 composition in the French idiom, the Jongen work is almost as conservative as Marshal Pétain, but Mata and Guillou play it for all it is worth. The Saint-Saëns is performed with broader strokes and much more excitement than in the Krivine/Matthes version (see under Denon above). The sound is just a wee bit more reverberant and hazy than I like, and the low notes of the organ are almost too powerful, but even so this is quite an audiophile showpiece, very clean and dynamic.

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Sergey Prokofiev: *Alexander Nevsky Cantata.* ***Dmitri Shostakovich:*** *Symphony No. 9 in E-flat Major, Op. 70. Dallas Symphony Chorus; Mariana Paunova, contralto; Dallas Symphony Orchestra, Eduardo Mata, conductor. DOR-90169 (1992).*

Mata is in his element here; these are absolutely first-rate performances, perhaps without that special touch of Russian "soul," but well-rehearsed, incisive, and idiomatic. The "Battle on the Ice" scene in *Nevsky* is awesome. The tuneful, catchy 9th is Shostakovich's "classical symphony" and is played here with the appropriate lightness. As for the sound, it's pre-20-bit Dorian but one of their absolute best, ever. I still insist that not all of the Craig Dory recordings in McDermott Hall have this kind of definition and just-right liveness. The bass drum alone is worth the price of admission (that's one of my favorite cliches, as you may have noticed).

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Maurice Ravel: *Piano Trio in A minor.* ***Cécile Chaminade:*** *Piano Trio No. 1 in G Minor, Op. 11.* ***Camille Saint-Saëns:*** *Piano Trio*

No. 1 in F Major, Op. 18. The Rembrandt Trio: Valerie Tryon, piano; Gerard Kantarjian, violin; Coenraad Bloemendal, cello. DOR-90187(1993).

The Ravel trio is a minor masterpiece; the Saint-Saëns is utterly charming; the Chaminade I could pass up without regrets. The Rembrandt Trio plays impeccably; they're fine musicians. The 20-bit recording, in the Troy hall as usual (but without Craig Dory) is perfection itself; the instruments are vivid, and the hall sound is not exaggerated. Demo quality.

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***Robert Schumann:** Piano Quartet in E-flat Major, Op. 47. **Johannes Brahms:** Piano Quartet in G Minor, Op. 25. The Ames Piano Quartet: Mahlon Darlington, violin; Laurence Burkhalter, viola; George Work, cello; William David, piano. DOR-90194 (1993).*

Another 20-bit recording of chamber music in the Troy hall, again without Craig Dory and again of the utmost refinement and palpable realism with just-right ambience. On top of it, this is lovely music (both works, even if I'm more of a Schumann fan than a *Brahmsianer),* and the Ames quartet plays very stylishly, with a fine combination of nuance and brio.

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"GRAND CONCERT! Vocal and Instrumental Music Heard in 19th Century America." D.C. Hall's New Concert and Quadrille Band. DIS-80108 (1991).

This is a release in the Dorian Discovery series, which includes licensed miscellanea not necessarily recorded by the Dorian team (in this case by Sony Classical). The D. C. Hall band is a weird antiquarian group specializing in recreating mid-19th-century American genteel/cornball (parlor, gazebo?) musical performances in the most authentic manner possible. It's quite amusing. There's a quintet of flute/piccolo, clarinet, violin, viola, and bass violin, plus a tenor—and what a tenor! His name is Kevin M'Dermott; he has a distinctly Irish tenor sound; and in sheer vocal quality I would rank him just a smidgen above John McCormack. What? Am I out of my mind? Not at all. I know a great voice when I hear one. Of course, "Come into the Garden, Maude!" is not the same test of vocal artistry as "// *mio tesoro,"* but I think if he sang Don Ottavio at the Met (assuming he could learn the role), the critics would be falling all over themselves. He has a much more beautiful voice than any of the aging Three Tenors, for example. At this point he seems to be the guiding spirit of the D. C. Hall group and maybe that's all he wants to do. And maybe I'm overreacting, so find out for yourself—if you can stand the almost comically banal music on this CD. One of a kind and very well recorded.

Erato

Without being considered an "audiophile" label, Erato consistently delivers as good sound as anyone—and, of course, they have the Time Warner clout when it comes to obtaining the best artists.

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***J. S. Bach:** St. Matthew Passion, BWV 244. The Amsterdam Baroque Orchestra; choruses and soloists; Ton Koopman, conductor. 2292-45814-2 (1992).*

An original-instrument, period-practice *Matthäus-Passion,* a bit on the cool side but effective nonetheless, with a nice sustained flow. The instrumental playing is better than the singing; the sound is extremely transparent.

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***J. S. Bach:** The Art of Fugue, BWV 1080. Marie-Claire Alain, organ. 4509-91946-2 (1992).*

The absolute pinnacle of contrapuntal composition (left in open score form by Bach but obviously intended for some kind of keyboard) is realized with great clarity here on a baroquely voiced Alsatian church organ built in 1975. Alain uses very colorful registration but wisely avoids the full organ. Her phrasing is chaste in comparison with that of a Jean Guillou, but Bach is probably better served that way. An authoritative, yet not pedantic, exegesis of this abstruse masterpiece, beautifully recorded and surprisingly listenable.

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***Johannes Brahms:** Symphonies Nos. 1-4; Tragic Overture; Academic Festival Overture; Variations on a Theme by Haydn. Chicago Symphony Orchestra, Daniel Barenboim, conductor. 4509-95191/2/3/4-2 (1993-94).*

Surprisingly sensitive, nuanced, and uneccentric conducting by Barenboim; world-class playing by the orchestra; absolutely lovely sound captured by the paradoxically named Larry Rock in the supposedly "impossible" Orchestra Hall—overall this turns out to be one of the more desirable Brahms symphony sets of the digital era. It's easy to nitpick even the greatest conductor's phrasing of every four bars of this music, but no one can call these performances routine or boring. One stupid boo-boo: in the third movement of Symphony No. 3, at 2:30, there is an intolerably bad splice that ruins the flow of the music.

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***Johann Strauss:** Waltzes and Polkas. Chicago Symphony Orchestra, Daniel Barenboim, conductor. 2292-45998-2 (1992).*

Unsurpassed melodist, master craftsman, master orchestrator—J, Strauss is underestimated as a "light" composer. But you have to play him with a light touch, otherwise the debonair Viennese magic is gone. Barenboim doesn't have the light touch; he leans into the music and drives it hard, on a tight rein. Even Erich Kunzel and the Cincinnati Pops (Telarc) are better, because they're looser, and that's not half the virtuoso orchestra the Chicago is. The sound itself is as good as in the Brahms set.

Harmonia Mundi

This is another label that strives for authenticity and serious musical values in their recording projects, rather than the exploitation of trendy glamour. Handel appears to be their favorite composer.

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***George Frideric Handel:** Messiah. Five soloists; Les Arts Florissants chorus and orchestra, William Christie, conductor. HMC 901498.99 (1993).*

Maybe if I were more of a period-practice purist, I would find this chamber-sized, elegantly played and sung performance (of the original Dublin score only) less wimpy and ineffective. This not the Handel who has thrilled millions over the centuries. The magnificence is missing. The earlier Harmonia Mundi *Messiah,* conducted by McGegan, was emotionally more satisfying—and of course much more complete. The recorded sound of this new version is, on the other hand, wonderfully transparent and timbrally accurate.

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***George Frideric Handel:** Radamisto. Ralf Popken, countertenor; Juliana Gondek, soprano; Lisa Suffer, soprano; et al.; Freiburger Barockorchester, Nicholas McGegan, conductor. HMU 907111.13 (1993).*

Handel's half-forgotten operas are an incredible treasure trove of marvelous music, and *Radamisto* is one of the greatest, even though this is the first commercially available recording of it—can you believe it? This beautifully sung and superbly conducted performance (and it isn't even McGegan own orchestra!) would deserve a feature review by a Handel specialist; let this dilettantish reviewer merely state that, if you like the early-18th-century kind of florid singing (and it ain't Puccini, paisan), then this is a must. The recorded sound is as limpid, palpable, and properly scaled as it always seems to be, regardless of the engineer, when Robina Young is the producer.

***George Frideric Handel:** Arias. Lorraine Hunt, soprano/mezzo-soprano; Philharmonia Baroque Orchestra, Nicholas McGegan, conductor. HMU 907149 (1989-91).*

Speaking of the florid Handel style of singing, if you want a sampler by one of the great practitioners, here it is. There is no better Handel singer than Lorraine Hunt; there is no more authoritative Handel conductor than Nicholas McGegan; there is no more convenient single-CD introduction to this wonderful music than this one. Highly recommended.

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***Gustav Mahler:** Symphony No. 1 in D Major (with "Blumine"). Florida Philharmonic Orchestra, James Judd, conductor. HMU 907118(1993).*

The Florida Philharmonic is the big surprise here; they play within a hairsbreadth of the world's great orchestras. A little more weight and plush in the strings, and they'd be right up there. After his outstanding Hoist *Planets* (Denon), James Judd was no surprise to me; he does an equally intelligent, sensitive, meticulous job with Mahler, making this one of the better Firsts of the digital era. Peter McGrath's recording is state-of-the-art—again no surprise. The audiophile community picked up on it right away.

Koss Classics

This audiophile label is a subsidiary of the Koss headphone company, an outfit that presumably knows good sound.

***Maurice Ravel:** Valses nobles et sentimentales. Other short works by Villa-Lobos, Nazareth, Scriabin, et al. José Feghali, piano. KC-1018 (1991).*

The piano sound here is of the life-size, in-your-face, in-your-room variety and absolutely perfect of its kind—fantastic dynamics, low distortion (another Larry Rock classical job). The young Brazilian-born pianist Feghali has a big technique (don't they all these days?) and a respectable understanding of' the idiom, but his competition in this music is just overwhelming.

London

Of the Cleveland Orchestra performances discussed below, all but one were recorded by John Pellowe in Severance Hall over a period of about a year and a half. Only the *Till* is a John Dunkerley taping. The audio quality in all instances is typical of the English Decca multimiked approach, achieving a big, clear, layered sound with excellent dynamics and strong bottom foundation. The stunning intimacy and impact of the best Telarc and Delos recordings, however (to name only two examples), isn't quite equaled. It's more of a generic and somewhat homogenized sound—but still transparent, musical, and highly listenable. A bit of edginess obtrudes at rare moments, especially on loud brass, perhaps because of too close miking.

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***Anton Bruckner:** Symphony No. 5 in B-flat Major. The Cleveland Orchestra, Christoph von Dohnányi, conductor. 433 318-2 (1991).*
***Anton Bruckner:** Symphony No. 6 in A Major. **Bach/Webern:** Ricercare. The Cleveland Orchestra, Christoph von Dohnányi, conductor. 436153-2 (1991-93).*

Bruckner's great, soaring themes and colorful orchestration almost play themselves, but coherence isn't his long suit. A conductor who can make it all hang together is way ahead of the game. Dohnányi has that ability, and the superbly clean, articulate, disciplined playing of the Clevelanders helps a great deal. No two Brucknerites will ever agree on the "best" interpretation of these problematic works, but I think these performances are right up there with the top contenders. At the very least, none are better played technically.

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Franz Liszt: *A Faust Symphony. Royal Concertgebouw Orchestra, Riccardo Chailly, conductor. 436 359-2 (1991).*

This is much greater music than most of the Liszt that you hear much more often, and Chailly gives it all the nuanced attention and expressive phrasing a Romantic masterpiece deserves. This kind of music is perfect grist for the great orchestra's mill, and they give it everything they've got. Interestingly, after my blanket comments above on the sound of the Cleveland recordings, John Dunkerley's excellent results here in a much better hall appear to have just about the same virtues and occasional faults. Decca blood is thicker than Amsterdam water.

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Gustav Mahler: *Symphony No. 4 in G Major. Dawn Upshaw, soprano; The Cleveland Orchestra, Christoph von Dohnányi, conductor. 440 315-2 (1992).*

There isn't nearly as much portentousness and breast-beating in the cheerful Fourth as in the other Mahler symphonies; that's why some people like it best/least of all. Dohnanyi responds to the relative simplicity and classical proprotions of the work; those who expect a lot of heaving and convulsions will be disappointed. The orchestral playing is absolutely superb, and Dawn Upshaw is a highly intelligent and musical interpreter of the last movement, even if vocally less than amazing.

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Richard Strauss: *Ein Heldenleben, Op. 40; Till Eulenspiegels lustige Streiche, Op. 28. The Cleveland Orchestra, Christoph von Dohnányi, conductor. 436 444-2 (1991-92).*

A very solid, impressive, flawlessly played performance of *Heldenleben* is diminished by the fact that the Reiner/Chicago version of 1954 is even better and almost as well recorded. ("The best is the enemy of the good," said Voltaire.) *Till* used to be something of a *spécialité de la maison* of George Szell in his Cleveland days, and it seems the orchestra is still good at it.

Mapleshade

This label, devoted mainly to jazz and blues, uses custom electronics and minimalist recording techniques to obtain a sound of astonishing transparency and presence, probably the best of its kind known to me. Their master tapes are live-to-2-track analog, recorded on a special machine with very extended frequency response—and as far as I'm concerned that's their tweaky privilege if the results are this good. I have picked three of their recent releases here almost at random; there's more, and just about all of it of super demo quality.

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"Masters from Different Worlds." Clifford Jordan, tenor and soprano saxophone; Ran Blake, piano; Julian Priester, trombone; etal. MS 01732 (1989).

The idea here is to contrast the basically mainstream Chicago postbop Jordan (who died recently) with the crazy avant-garde Blake. The result is some very interesting sounds and damn fine jazz.

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"Portraits in Ivory and Brass." Jack Walrath, trumpet; Larry Willis, piano; with Steve Novosel, bass. MS 02032 (1992).

This is pretty sophisticated modern jazz played by two classically trained musicians. It may be too far out for some but no more than, say, Sonny Rollins. To me it sounds just right.

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"Highways of Gold." Harvey Thomas Young, vocals and acoustic guitar; with Junior Brown, guit-steel and pedal steel. MS 02252 (1992-93).

This is the kind of singing and guitar playing you might hear in a Texas bar—if you're lucky. Singer-songwriter Young has an appealing style, but the locally more famous "guitar legend" Junior Brown is very laid-back on this disc, probably because he'd steal the show if allowed to cut loose.

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MusicMasters Classics

A label that uses Max Wilcox to produce their major releases, such as the one below, is already on the right track in my book.

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J. S. Bach: *The Well-Tempered Clavier, BWV 846-869, Book I. Vladimir Feltsman, piano. 01612-67105-2 (1992).*

Feltsman is a brilliant technician and also somewhat willful in his phrasing of Bach. The result ranges from the sublime to the irritating. The piano sound is simply gorgeous.

Reference Recordings

This label confesses the audio gospel according to Saint Johnson (Keith O.).

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"Trittico." Dallas Wind Symphony, Frederick Fennell, conductor. RR-52CD (1992).
"Pomp & Pipes!" Paul Riedo, organ; Dallas Wind Symphony, Frederick Fennell, conductor. RR-58CD (1993).

Both of these programs of short, mostly 20th century, mostly showy pieces, effectively conducted by the ancient Fennell, were recorded in McDermott Hall and are intended to be played back with HDCD decoding, which I don't have yet as of this writing. Undecoded the recordings have sensational dynamic range, awesome bass, very nice ambience and dimensionality, but I've heard greater transparency and finer detail on other RR CDs. Whether HDCD is an advancement in accuracy or just another processor remains to be seen.

Telarc

What other audiophile-oriented label has as many good artists as Telarc? None.

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J. S. Bach: *Brandenburg Concertos. Boston Baroque, Martin Pearlman, director. Nos. 1, 2, & 3: CD-80368 (1994). Nos. 4, 5, & 6: CD-80354 (1993).*

As authentic, propulsive, and convincing as any period-instrument performance in the catalog—and better recorded. What else is there to say?

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Ludwig van Beethoven: *String Quartet in E-flat Major, Op. 74 ("The Harp"); String Quartet in F Minor, Op. 95 ("Serioso"). Cleveland Quartet: William Preucil and Peter Salaff, violins; James Dunham, viola; Paul Katz, cello. CD-80351 (1991-92).*

The utmost finesse, as against the powerhouse school of Beethoven playing, characterizes these performances, which are at least as successful artistically as the previous two CDs in the series. The recording is pure silk and of superb transparency. (The producer was Judy Sherman, Max Wilcox's ex.)

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Manuel de Falla: *La Vida Breve. Alicia Nafé, mezzo-soprano; Antonio Ordonez, tenor; May Festival Chorus, Robert Porco, director; Cincinnati Symphony Orchestra, Jesus López-Cobos, conductor. CD-80317(1992).*

Beautifully sung, idiomatically conducted, and superbly recorded performance—definitely the pick of the digital era in this short, uneven, but extremely vital early opera of the composer. Killer flamenco passages.

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Alexander Glazunov: *The Seasons, Op. 67 and 67a; Scènes de Ballet, Op. 52. Minnesota Orchestra, Edo de Waart, conductor. CD-80347(1993).*

The scrumptious Russian ballet music is just one attraction here; another is the verve of the Minnesota players; and still another the audio quality—unusually clean, smooth, and natural even for Telarc. Is it the new and different (for them) venue or the 20-bit A/D processing? Only Jack Renner knows for sure.

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W. A. Mozart: *Cost fan tutte. Felicity Lott, Marie McLaughlin, Nuccia Focile, sopranos; Jerry Hadley, tenor; Alessandro Corbelli, baritone; Giles Cachemaille, bass-baritone; Edinburgh Festival Chorus; Scottish Chamber Orchestra, Sir Charles Mackerras, conductor. CD-80360-A/B/C (1993).*

A very important release, deserving of an extended feature review a la David Ranada. In future years this may possibly be ranked as the digital era's equivalent of the 78-rpm era's Busch/Glyndebourne 1935 recording. Mackerras conducts with a combination of easy flexibility and disciplined control in perfect dynamic balance, and the voices are all fresh and lovely. Of course, the star is still Mozart. (Stravinsky played this music over and over again on his phonograph while composing *The Rake's Progress.)* The 20-bit recording is absolutely beautiful in texture but presents a relatively small soundstage, perhaps deliberately, to suggest a small, intimate theater.

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Sergei Rachmaninoff: *Symphony No. 3 in A Minor, Op. 44; Symphonic Dances, Op. 45. Baltimore Symphony Orchestra, David Tinman, conductor. CD-80331 (1994).*

To me, this CD proves two things. One, that Rachmaninoff composed his best music in his 60s. Two, that Jack Renner can occasionally surpass his normal high standard in recorded sound. The string tone, dynamics, and rounded solidity of this recording are extraordinary. And, yes, the Baltimore has become a very fine orchestra.

Teldec

See also the Wagner *Ring* review on page 52.

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Béla Bartók: *Concerto for Orchestra; The Miraculous Mandarin (Suite for Orchestra); Deux Images. Philharmonia Orchestra, Hugh Wolff, conductor. 9031-76350-2 (1993).*

The great acoustics of Watford Town Hall, superb playing by a world-class orchestra, sensational recording by Teldec's German team—these are the strengths of this release. As for Wolffs conducting, his inauthentically slow tempo in the second movement of the Concerto tells the story: too careful, not incisive enough, not at all like Reiner or Solti. (Call me a Hungarian chauvinist.)

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Gaetano Donizetti: *Lucia di Lammermoor. Edita Gruberova, soprano; Neil Shicoff, tenor; et al; The Ambrosian Singers; London Symphony Orchestra, Richard Bonynge, conductor. 9031-72306-2 (1991).*

I agree with Toscanini about *Lucia: "Che bell' opera!"* Never mind the coloratura shenanigans; listen to the melodies, the ensemble writing, the lyricism alternating with drama—it's good stuff. Here is the same pair in the leading roles that I liked so much in the Teldec *Traviata:* Gruberova has touches of greatness (vocally if not dramatically) and Neil Shicoff has a still fresh, unabused, beautiful tenor voice. Bonynge's conducting could be more dynamic but he keeps producing lovely sounds, and that's what this opera is about. The recording is perhaps a bit constricted in the busy moments but always clean.