The Audio Critic®

Issue No. 24
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Plus lots more test reports, all our regular columns, letters to the Editor, and a nice assortment of CD reviews.

Contents

Issue No. 24 Spring 1997

Editor and Publisher	Peter Aczel
Contributing Technical Editor	David Rich
Contributing Editor at Large	David Ranada
Technical Consultant	Steven Norsworthy
Technical Consultant (RF)	Richard Modafferi
Columnist	Tom Nousaine
Cartoonist and Illustrator	Tom Aczel
Business Manager	Bodil Aczel

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From the Editor/Publisher:

Once again we have broken our own lateness record. It is now fairly obvious, after numerous Band-Aid attempts at improvement, that we are never going to have a normal publishing schedule in our present mode. What is that mode? Basically a one-man operation. One man does all of the laboratory work (except RF), nearly all of the writing, all of the editing, all of the desktop publishing, all of the administrative work (except subscriptions). Yes, there are some contributors, but their copy is at least as time-consuming to edit as writing original articles. One advantage of the system is that there is hardly any overhead, so we cannot possibly go bankrupt like some other audio publications. We are here to stay. But we must become part of a larger organization, whether an expanded version of our present one or some already existing outside one. That is what we are currently working on. Do not worry about our editorial autonomy, however; we are not going to give it up as part of some deal.

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Letters to the Editor

Unlike those enormous electronic toilet walls on the Internet on which any lowlife can write his name and effusions, this is definitely a "moderated" column. We welcome intelligent and relevant commentary, especially when typed or word-processed. Address all editorial correspondence to the Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951-0978.

The Audio Critic:

There is a common misunderstanding of sound imaging, perpetuated in Daniel Sweeney's article "Twice Shy: On Reencountering Multichannel Music Formats" in Issue No. 23, which I would like to address. Sweeney bemoans the presence of "interaural crosstalk" in surround sound systems in general, and Ambisonics in particular, as if there is something inherently wrong about each ear hearing all the loudspeakers. This fallacy can be traced right back to a common belief that one of the defects of stereo reproduction is that each ear hears *both* loudspeakers, and that the ideal would be that the left-channel signal be heard only by the left ear, and the right-channel signal only by the right ear. Such a situation (which can be achieved by the use of headphones, or crosstalk-cancelling signal processing feeding the loudspeakers) would represent *binaural* reproduction and would require a binaural ("dummy head") recording as the source. But Sweeney is not discussing binaural sound, and both stereo and Ambisonics are predicated on the existence of this crosstalk.

Ambisonics goes further than stereo in that what it does (to first order) is to sample the acoustic field in such a way that the *combination* of the signals from all the loudspeakers in the array produces, in a region of space around the center of the array, a reconstruction of the original acoustic wave field (both travelling- and standing-wave components). If a listener puts his or her head in this sound field, then, because the wave fronts are similar to the original, the perception of directionality and space should correspond to the original too. It is a "wave front reconstruction" scheme in the small. The ear signals (crosstalk and all) will be correct if the reconstructed wave fronts are correct. This is just like natural hearing. Increasing the number of loudspeakers in Ambisonics (each fed its correctly decoded signal) increases the accuracy of the reconstruction and the region over which it holds up. The interaural arrival-time differences also correspond to natural hearing in Ambisonics.

All these aspects are correctly captured to first order by Ambisonics. It is incorrect to reason that, because the Calrec "Soundfield" microphone uses a single-point multichannel pickup, it cannot take into account the spacing of the human ears. There is no logical connection between the two: if the wave fronts are right, the listener *must* hear correctly. It's as simple as that!

Sincerely,
Stanley P. Lipshitz
Department of Applied Mathematics
University of Waterloo
Waterloo, Ontario, Canada

As our readers know, I don't shrink from confrontation or debate, but even so I have two unbreakable rules: (1) don't mess with the Lone Ranger and (2) don't argue with Stanley Lipshitz.

Dan Sweeney is braver than I am, so I'll let him answer the above letter, as well as the following two on the same subject. Obviously, he dropped his left just a little bit when he tried to dispatch Ambisonics in the first round, but I think he is still well ahead on points. If I didn't think so, I wouldn't have asked him to do a sequel on multichannel formats for this issue.

—Ed.

The Audio Critic:

In his article "Twice Shy: On Reencountering Multichannel Music Formats," Daniel Sweeney, Ph.D., reveals that he does not hold a doctorate in surround-sound technology. *[It was I who inserted his degree, earned in a totally unrelated discipline, out of sheer snobbery and elitism.—Ed]* Permit me to correct his misstatements specific to Ambisonics. Dr. Sweeney begins his commentary by comparing the fate of Ambisonics to that of the California condor. I would hasten to add, however, that Ambisonics has yet to become extinct because, like the condor, it is an elegant evolution!

Dr. Sweeney's most glaring misstatement is his claim that Ambisonics "is in a real sense the last survivor of the quadraphonic experiment." The most elementary understanding of Ambisonics belies that characterization. Quadraphony is nothing more than stereophony doubled; Ambisonics, on the other hand, is a mathematically proven system predicated upon the encoding of the *direction* of sound as it arrives at a *single* point in space from every azimuth and elevation—hence the name "ambi," which means "around"—and the reproduction of that sound field based upon psychoacoustic theory. In quad, 4 microphones oriented at 45, 135, 225, and 315 degrees feed 4 "transmission channels" (the channels which convey the sound field to the listening room), which in turn feed each of 4 respective loudspeakers located at 45, 135, 225, and 315 degrees. According to Ambisonic theory, however, it is imperative that there be *fewer* transmission channels than loudspeakers! In the case of full horizontal surround (called "pantophony"), 3 transmission channels are sufficient to reproduce the sound field via upwards of 4 loudspeakers. For full spherical surround (called "periphony"), requiring upwards of 6 loudspeakers, only 4 transmission channels are necessary. Furthermore, unlike quad, which by definition comprises only 4 loudspeakers, there is theoretically no limit to the number of loudspeakers that can be employed in Ambisonic reproduction. Four speakers are the absolute minimum for pantophony, while at least 6 are needed for periphony. Decoders are currently available which provide 5-, 6-, and 8-channel pantophonic playback. (Decoders can be daisy-chained to allow 16 loudspeakers for large installations.) In addition, Ambisonic decoders are designed to accommodate any geometric layout of loudspeakers, e.g., a square, rectangle, pentagon, hexagon, octagon, etc. In this respect, Ambisonics differs from all other current multichannel surround systems that are designed to employ a fixed number of loudspeakers, such as Dolby 5.1. Because actual sources of sound are more realistic than phantom ones, the more loudspeakers, the merrier.

Dr. Sweeney is quite confused about the various Ambisonic formats. B-Format is *not* the consumer format as he states but, rather, the professional format. To record a sound field, a specially designed microphone called the "Soundfield" is generally employed. The Soundfield contains a tetrahedral array of 4 subcardioid condenser capsules whose respective outputs (called "A-Format") are added and subtracted to produce the following acoustically equivalent microphone signals: W, X, Y, and Z, where W is an omni pattern, X is a forward-facing figure-eight pattern, Y is a side-facing figure-eight pattern, and Z is an upward-facing figure-eight pattern (collectively called "B-Format"). Thus both incidents of sound—pressure and direction—are sampled at a single point in space. (I note in passing Dr. Sweeney's criticism of Ambisonic's failure to record time-of-arrival localization cues. This criticism is misplaced, for a space can be encoded to B-Format without the use of a coincident microphone such as the Soundfield; that is, B-Format can be derived from a multiplicity of spaced microphones. However, I would refer him and other interested readers to an Audio Engineering Society article by noted mathematician Dr. Stanley Lipshitz entitled "Stereo Microphone Techniques," wherein he argues that any recording technique departing from the single-point approach as idealized by the Soundfield is a fundamentally flawed, albeit sonically pleasing, representation of space. (See *Journal of the Audio Engineering Society* 34 [September 1986]: 716-44.)

What most people have heard of Ambisonics is the 2-channel consumer version called "UHJ" or "C-Format" and made somewhat popular by Nimbus Records as well as other European companies (Finlandia, Ondine, Unicorn-Kanchana, etc.). Like Dolby Pro Logic, UHJ is a matrixed version of the discrete multichannel B-Format. While UHJ is compatible with stereo reproduction, it is designed to be decoded Ambisonically; however, it is no substitute for the sonically superior B-Format. UHJ is a thing of the past, since the delivery of B-Format into the home is now technologically feasible with the advent of the multichannel DVD format. The raison d'etre of Ambisonics is 360-degree localization: the musicians are heard before you, the audience behind, and ambience and reverberation all around. It is ironic that the high-end press should rail against digital because it fails to encode the sonic information between each ¼8,000-second sample, yet they entirely ignore the fact that stereo itself is a grossly lossy system inasmuch as five sixths of the sound field is forsaken. Not only does Ambisonics reproduce a space with high fidelity, but the timbre of the instruments so decoded is not compromised by the effect of comb filtering produced by the summing of the direct and indirect sound fields, as is the case with stereo. This fact is rarely appreciated by those in the high end defending stereo as the end-all-be-all.

...When more recordists experiment with surround sound, Ambisonics, I am confident, will be found to provide the most convincing and palpable illusion of "being there." (See Furlong, D. J., "Comparative Study of Effective Soundfield Reconstruction," 87th Convention of the AES, New York, NY, October 18-21, 1989: Preprint 2842.) Before he dismisses it, Dr. Sweeney would do well to give Ambisonics a full listen...Dr. Sweeney is correct in pointing out that there is a worldwide fraternity of "True Believers..."

Very truly yours,
Jeffrey Silberman
The Surroundworks™
Mill Valley, CA

I feel moved by Dan Sweeney's "Twice Shy" article in your Issue No. 23 to point out that it is full of errors. I'm afraid his "wide-ranging knowledge of the multichannel scene" with which you preface the article has a number of "holes in the middle" in it.

I found Mr. Sweeney's article in *Audio* on the same subject to have some very strange and important omissions in it—such as no mention whatever of the ARA group's activities or HQCD—but the *Audio Critic* article is much more blatant. Time doesn't permit my going into each one, and I'm sure others closer to Ambisonics than myself will detail all of them at length, but to name just a few:

Ambisonics has little or no connection with any Japanese developments. UMX is something else entirely. Mr. Sweeney has B-Format and UHJ format confused, or else he appears to not know about UHJ at all. All the commercial recordings issued so far in Ambisonics on Nimbus, Unicorn, etc., have not been B-Format at all but UHJ, which is the two-channel version. B-Format is three or four channels.

The "two significant factors in the recording/playback process" that have been ignored according to Mr. Sweeney are not at all significant to the results with Ambisonics, but I'll let others fill in the technical specifics on that matter. While it is true that many UHJ recordings when heard via two stereo speakers sound overly reverberant, that can be remedied without necessarily requiring a dedicated Ambisonic decoder. The simplest passive Hafler circuit can extract the surround information and feed it to surround speakers, which instantly gives a drier, more accurate acoustic to the venue and places the performers with quite precise localization (though not as perfect as with an Ambisonic decoder).

As for your review in the same issue of the Onkyo TX-SV909PRO receiver with Ambisonics decoder, there was a good reason why there was "little or no interest in its Ambisonic decoding capability." Onkyo didn't promote it at all, and on top of that they left out the "Stereo Enhance" function that other Ambisonic decoders have had. This creates the most natural and musical surround to be had from any standard stereo recording (surpassed only by an enhance function in Meridian's 565). And it means owners of the 909 wouldn't have had to purchase special Ambisonic CDs in order to experience much of the surround magic.

I don't know why Mr. Sweeney would think Ambisonics to be a "horrible name," but it certainly has had its share of horrible promotion, marketing and distribution. One of the many counts against it has been the poor-quality circuitry of most of the consumer-level decoders such as the Minims. However, there is an excellent professional Ambisonics decoder available for around $3000, which handles both UHJ and B-Format, and competes with the Meridian, Lexicon, and other highest-end processors.

I have had various varieties of surround sound in my listening room/studio for 35 years now, and to my ears UHJ Ambisonics is hands down the best that can be delivered with two channels, and B-Format Ambisonics the best if more channels are available. Pitted against Dolby Pro Logic and using music sources rather than soundtracks, there's no contest whatever. Only true binaural via headphones surpasses it in sonic realism.

Sincerely,
John Sunier
Audiophile Audition
Ross, CA

Dan Sweeney replies to Stanley Lipshitz, Jeffrey Silberman, and John Sunier:

In regard to Dr. Lipshitz's letter, I must admit I erred in supposing that Ambisonics cannot capture time-of-arrival cues. In regard to his other point on crosstalk cancellation schemes (such as Carver Sonic Holography, for example), I have read his previous remarks on the subject. I remain unconvinced. Loudspeaker arrays add an overlay of time and amplitude differences on top of those in the recording. I am at a loss to understand how those wouldn't make a difference—wouldn't contaminate the data, so to speak. Incidentally, I am perfectly well aware that binaural recording avoids the problem, and I think that that is a strong point in its favor.

For those who are unacquainted with the specifics of the discussion, I should mention that Dr. Lipshitz asserts in his 1986 AES article that classic minimalist stereo techniques take into account the interaural crosstalk produced by pairs of loudspeakers, thereby obviating the need for crosstalk cancellation schemes. But how precisely do they take that into account? That wasn't explained in the article, and frankly I don't see it, though I might be missing something. By the way, an extended discussion of the crosstalk problem is included in Durant Begault's *3D Sound for Virtual Reality and Multimedia,* a recent publication dealing with sound localization theory and practice, and I would refer interested readers to that text. One might also review Ralph Glasgal's privately printed *Ambiophonics,* which contains an extensive review of the literature. If I am in error here, I am not alone.

* * *

Mr. Silberman's letter manifests precisely that carping tone that led me to give such short shrift to Ambisonics in the first place. Any mention of the subject is sure to summon up legions of ghosts ready to refight battles that were lost long ago. At the risk of reanimating those ghosts, let me state very plainly and unequivocally the obvious: Ambisonics is dead—dead as a boot. How many new titles were released this year, or last year, or the year before? How many decoders are on the market? Ambisonics makes HDCD look like a howling success. It had its day in court fifteen years ago, and the community of recording engineers rejected it decisively. And unlike vinyl or vacuum tubes, Ambisonics isn't enjoying even a modest revival.

As for Ambisonics's relationship to quad, that is real. Ambisonics grew out of the Nippon Columbia UMX system back in the middle seventies, and both UMX and Ambisonics fought it out with SQ, QS, and the rest for market share—and failed even more spectacularly, I might add. I will say, however, that Mr. Silberman's description of Ambisonics is generally accurate. Ambisonics is appealing on conceptual grounds, and it does differ from the other multichannel music formats of the seventies in that the channels on the recordings do not coincide with speaker feeds. But so what? Like SQ and QS, Ambisonics was (note I said *was* because it is virtually dead) a low-separation, matrixed multichannel format depending on amplitude and phase panning to create phantoms in all quadrants. Yes, it could support any number of speakers, but separation went down, down, down when speakers were increased. Ambisonics supporters claim that doesn't matter, but the recording community sure thought otherwise.

And that's all I have to say. Let Silberman continue to flog Ambisonics on the Internet. History has passed it by.

* * *

Regarding John Sunier's letter, point by point:

When I wrote the article in *Audio,* the ARA had published little. In any event, what they have published to date has little to do with recording techniques other than the fact that they endorse—you guessed it—Ambisonics.

UMX is cited as predecessor to Ambisonics in a number of early articles on the subject.

I do know about UHJ, though Sunier and others are correct: I misidentified B-Format with UHJ. Incidentally, two channel UHJ is all that is available, or ever has been available, commercially. B-Format has never been anything but a mastering format. I disagree strenuously with Sunier and others that UHJ sounds wonderful. To me it's unlistenable, which is one of the reasons I grow impatient with Ambisonics boosters. Until DVD offered the possibility of a commercial B-Format, most of them insisted that UHJ was damned near perfect. Such pronouncements just don't accord with my experiences, and I'm not alone there. The fact is it failed in the marketplace, and if that's due to botched promotion, then the responsibility falls squarely on the shoulders of its boosters, not me. Hey, I didn't kill it, guys, you killed it. But then, as Oscar Wilde famously noted, we always kill the thing we love.

As for the localization cues ignored by Ambisonics theory not being significant, if they exist, how could they not be significant? Let's create a Procrustean model of human perception based on a twenty-year old theory and ignore everything since.

I have a proposition to make to all you Ambisonicists out there who are reading this and are preparing to write more letters or flame me on the Internet. Instead of further grousing, pool your resources and start issuing B-Format recordings on DTS disc. That will give you up to five totally discrete channels to work with and 20-bit resolution, at the cost of four-to-one compression. If the B-Format is everything you say it is, then the market will embrace it in that form.

Won't it?

Daniel C. Sweeney
Burbank, CA

The Audio Critic:

Thanks for the kind words in your "Hip Boots" remarks. Calling us a "pleasant little journal" and acknowledging that at least some of what we do has merit is good enough for me. You could have been much nastier.

We will never be as single-minded as *The Audio Critic*, but I do hope that we will be seen as helping audiophiles and music lovers to spend less and get more, which is a goal that I believe our two publications have in common, even though we go about things differently in some respects.

At any rate, I am not writing to grouse about being termed a "lightweight" when my doctor is threatening to enroll me in Weight Watchers; rather, I want to point out that one of the things I really enjoy about *The Audio Critic* is the amount of space you devote to music reviews and the perspective you offer on recordings. We need more CD reviews in future issues, Peter, not fewer—your latest issue seemed a bit skimpy (lightweight?) in its coverage of music.

It always amazes—and disappoints—me that I get so few letters from readers about music, in comparison with the letters I get taking me to task for not being a "believer" in cable differences and other such subjects. I'll bet you have felt the same way from time to time.

Keep up the good work, Peter, and keep putting plenty of CD reviews in each issue. I really appreciate your efforts, and I wish you the best in getting your publishing schedule on track.

All my best,
Karl W. Nehring
Editor
The Sensible Sound
Ostrander, OH

Graciousness will get you nowhere, Karl. We do indeed go about things differently. The main problem I have with your publication is, as I said, your policy of "equal time" for scientific truth and golden-eared nonsense, and you seem to be unwilling to address that issue.

If you want more CD reviews, read Fanfare *and* American Record Guide; *we are certainly no better than they are in that area, whereas when it comes to credible audio reviews...hey, don't let me get started on that subject.*

As for our publishing schedule, we need a small full-time core staff, not good wishes. Believe me, I'm working on it. And thanks for the kind words.

—Ed.

The Audio Critic:

I am pleased that your magazine is still being published. I have subscribed to two of the new magazines—*Fi* and another glossy one whose name I have forgotten. I canceled both subscriptions. They both seemed to like the high end when it was high-priced, and they had such a strong subjectivistic "placebo-effect" driven bias that I could not afford, or trust, their judgments.

I look for quality that is value. I want the product that represents the inflection point of the asymptotic curve linking cost and performance. I want to know where the point is when an additional dollar spent results in very little increase in sound quality, but when spent elsewhere can make a big difference. Your magazine comes closer to addressing these issues than any of the others.

I appreciate your intellectual paradigm for the evaluation of audio.

I do subscribe to *Stereo Review* to maintain an overview of the field, and I do read *Stereophile* because they do a good job of keeping up with the technical issues and technology of audio. However, I almost never read their product reviews—unless it is the rare item that I am interested in which also represents good value.

Sincerely yours,
James M. Larson, M.D.
San Diego, CA

You clearly have a good, and much appreciated, handle on where we 're coming from, doctor—but Stereophile *as a reliable source of information on audio technology? Read David Rich's review of their technical editor's book (page 82). Read it and weep.*

—Ed.

The Audio Critic:

There has been a lot of hot chatter on the E-mail circuit over the past couple of months about the Steve Maki and Steve Zipser challenge in Miami. I thought you would appreciate a complete recount of the events. Zipser, a high-end salon owner, had issued a challenge that he would pay the airplane fare of any interested party who wanted to see him prove he could hear the differences between amplifiers.

On Sunday afternoon, August 25th, Maki and I arrived at Zipser's house, which is also Sunshine Stereo. Maki brought his own control unit, a Yamaha AX-700 100-watt integrated amplifier for the challenge. In a straight 10-trial hard-wired comparison, Zipser was only able to identify correctly 3 times out of 10 whether the Yamaha unit or his pair of Pass Laboratories Aleph 1.2 monoblock 200-watt amplifiers was powering his Duntech Marquis speakers. A Pass Labs preamplifier, Zip's personal wiring, and a full Audio Alchemy CD playback system completed the playback chain. No device except the Yamaha integrated amplifier was ever placed in the system. Maki inserted one or the other amplifier into the system and covered them with a thin black cloth to hide identities. Zipser used his own playback material and had as long as he wanted to decide which unit was driving the speakers.

I had matched the playback levels of the amplifiers to within 0.1 dB at 1 kHz, using the Yamaha balance and volume controls. Playback levels were adjusted with the system preamplifier by Zipser. I also determined that the two devices had frequency response differences of 0.4 dB at 16 kHz, but both were perfectly flat from 20 Hz to 8 kHz. In addition to me, Zipser, and Maki, one of Zip's friends, his wife, and another person unknown to me were sometimes in the room during the test, but no one was disruptive and conditions were perfectly quiet.

As far as I was concerned, the test was over. However, Zipser complained that he had stayed out late the night before and this reduced his sensitivity. At dinner, purchased by Zipser, we offered to give him another chance on Monday morning before our flight back North. On Monday at 9 a.m., I installed an ABX comparator in the system, complete with baling-wire lead to the Yamaha. Zipser improved his score to 5 out of 10. However, my switchpad did develop a hang-up problem, meaning that occasionally one had to verify the amplifier in the circuit with a visual confirmation of an LED. Zipser has claimed he scored better prior to the problem, but in fact he only scored 4 out of 6 before any difficulties occurred.

His wife also conducted a 16-trial ABX comparison, using a 30-second phrase of a particular CD for all the trials. In this sequence I sat next to her at the main listening position and performed all the amplifier switching functions according to her verbal commands. She scored 9 out of 16 correct. Later another of Zip's friends scored 4 out of 10 correct. All listening was done with single listeners.

In sum, no matter what you may have heard elsewhere, audio store owner Steve Zipser was unable to tell reliably, based on sound alone, when his $14,000 pair of class A monoblock amplifiers was replaced by a ten-year old Japanese integrated amplifier—in his personal reference system, in his own listening room, using program material selected personally by him as being especially revealing of differences. He failed the test under hard-wired no-switching conditions, as well as with a high-resolution fast-comparison switching mode. As I have said before, when the answers aren't shared in advance, "Amps Is Amps" even for the Goldenest of Ears.

Tom Nousaine
Cary, IL

I was the one who asked Tom, our columnist, to write up this information in letter form, just for the record, in anticipation of distorted versions of the story. Both he and I knew all along, of course, that all such challenges by Golden Ears are unwinnable, but there are still some wide-eyed audiophiles out there who haven't received the word. Since anyone who has sampled Zipser's foul exudations on the Internet must realize that the man is a pathetic loser, it should come as no surprise that he lost his hopeless challenge, just like all others who have tried.

—Ed.

The Audio Critic:

I am a happy subscriber and appreciate your efforts to demystify the reproduction of music. *The Audio Critic* has been a needed voice of late in the mystic wilderness the High End has become. My system choices have been constructively influenced by your publication (Carver "Amazings" and Sunfire amp, for example—thanks!). After reading your [Peter Aczel's] letter to the editor in the November 1996 *Audio,* I have a few thoughts on the subject of ABX testing for your consideration. I hope you welcome these ruminations as I intend them: a rational attempt to understand a possible weakness in our current level of understanding. They came about as a result of asking the question: if ABX testing is, as many listeners feel, inherently flawed, a rational reason would exist—what would it be?

I have personally experienced the diminishment of audible differences that matching output levels introduces into sighted A/B comparisons, and then the further elimination of differences introduced by making the same test blind. It's quite an education for an open-minded audiophile to do matched-level blind comparisons for the first time. After reading and thinking about both sides of this debate for years, it occurred to me where the flaw in matched-level blind comparisons may lie. Certainly not in matching levels, or any other of the methodologies you refer to in the aforementioned *Audio* letter—save one: blind listening conditions.

The need to eliminate the so-called "placebo effect" in critical listening is clear, but I have begun to question whether it can be done with no diminishment to perceptive ability. Consider that the term "placebo effect" originally refers in medical research to the effect unconscious processes have on the physical body; that is, psychological cause, mechanistic effect. Under those conditions the double-blind test protocol proves invaluable. But in listening tests, blind conditions have been used to eliminate delusional thoughts arising from the unconscious. This is accomplished by disallowing assimilation to the conscious (and hence, unconscious) which component is presently listened to. The question is this: can we do this with no deleterious effect? Isn't perception a process as likely enhanced as damaged by the conscious/unconscious linkage? Doesn't it seem a self-evident notion that, in fact, when any perceptive activity is at its most acute, unconscious processes invariably play an integral and enhancing role? Have we, in a well-intentioned attempt to eliminate the delusional component in critical listening, thrown the baby out with the bath water?

In simplest terms, perhaps hearing acuity is improved when we know what we're listening to. Critical perception of audible cues seems improved, perhaps due to memory enhancement in long-term comparisons. (Diminishment to the effect of memory may explain why some listeners characterize blind tests as "stressful" and "confusing.") If this is eventually shown to be true, we move backwards somewhat in our belief in rigorously established quality differences with blind comparison tests, since we then appreciate there may be no precise way to differentiate negative and positive unconscious effect on perception. Even so, we wouldn't be back to square one: we know that ABX comparisons give more information than not. The magnitude of audible differences falling below the ABX detection threshold are comparatively small. But even these magnitudes might explain the differences listeners, nearly universally, have disappear under blind conditions. The relative importance assigned to those differences in aesthetic terms would be back in the subjective realm, and of course worthy of investigation and debate.

In summation, I think we should begin to question the assumption that blind comparison testing is predicated upon: that the inadequately termed "placebo effect" can only have negative impact on the activity of critical listening. Perhaps a few parallels that might work to illustrate the general type and magnitude of phenomena in question is that of fuzzy intelligence in recent powerful digital programs, or in audio, dither improving the audibility of very low-level signals in digital processing. Subtle, perhaps—important, perhaps.

Dave King
New York, NY

I am publishing your letter here only because I have never heard a more desperately convoluted expression by an obviously intelligent and sincere audiophile of the same old basic fallacy, which more primitively stated goes: "Since we all know that the Krell sounds better than the Pioneer, what's wrong with all those

ABX tests that cover up such an obvious truth?" I'll call it the Harley Fallacy, a classic case of petitio principii*—taking for granted the truth of something that remains to be proved. The fact is that we do not all know that the Krell sounds better; only the brainwashed among us know. Read your own words carefully—you are assuming up front that what "many listeners feel, " what they "nearly universally " perceive, is true. Wrong! Where's the proof?*

In addition to the Harley Fallacy, you also seem to be laboring under a misapprehension of what goes on in a properly conducted ABX test. It is not like a blind A/B test at all. You are at all times allowed to listen to fully identified A and fully identified B, "delusional components" and all. When you switch to X, you are not asked to evaluate its sound, with or without delusions. All you are asked is to match its sound to A or B. You can switch back and forth between A and X, or B and X, as many times as you like. Does X match A, or does it match B? It's like matching a color swatch to the paint on two different walls. Does it match or doesn't it? Where does delusion come in?

It is true, of course, that your mental image of the beautifully engineered and gorgeously packaged Krell—your delusion, if you will—represents "value added. " I would never stop an orthopedic surgeon, or a hedge fund manager, from buying the Krell. It will make him feel more important. (It will also look more handsome on his shelf and probably last longer.) Still, the only way to prove that the Krell "sounds better" than the Pioneer is to identify it by its sound alone, at matched levels, without peeking. Why can't you live with that?

—Ed.

The Audio Critic:

I am sending you two reprints from *Science* and *Nature* that discuss "stochastic resonance." You have probably heard of this already. The latest one from *Science* has a section under the subhead "Electric results," the gist of which is summarized in the quote: "...the more noise, the greater the membrane's ability to sense the field." After reading the *Nature* article I made a copy to send to you but just didn't get it done. Then the *Science* article came out and reinforced the ideas I discuss next.

Our ears must work much the same way as the shark organs that sense electric fields, except in our ears hair cells are stimulated mechanically. According to these articles, if there is some background noise we are probably able to hear minute sounds better than in a dead quiet background. Quite possibly the reason for vinyl's "superior sound" is that the vinyl surface provides this background noise. Perhaps a live concert's sound is enhanced because of soft noise coming from the audience (and because it's live). I once read in *Stereophile* (sorry) just after CDs came out that one editor, I forget who, thought CD sound might differ from vinyl sound perhaps because vinyl's background noise enhanced the sound—so he added vinyl noise to the CD sound and thought it now rivaled vinyl's. But he had no scientific grounds for this idea, nor did he try in any way to test the notion further (as usual), or ask others to listen.

I would just like to know what you and your colleagues think about the role of stochastic resonance, record and tape hiss, the relative lack of all background noise on CDs, and how we normally hear sound. Has the dead quiet background sterilized CDs?

Don't tell too many people about this, otherwise we shall have to buy noise generators to fix our hi-fis.

Phil Brandt
Professor
Columbia University
New York, NY

"Stochastic resonance " is becoming a buzzword, but I knew well before the digital era that a small amount of white noise added to the signal can create an impression of greater spaciousness and transparency. I always believed this to be a psychoacoustic gimmick rather than the extraction of actual additional information, but then the whole subject is far from being my long suit.

I don't quite see how vinyl's "snap, crackle, and pop " could possibly reveal, rather than cover up, low-level information, but what you say about the soft background noise of a live musical event may very well be true. I know this: when such residual hall noise is used by the producer for the pause between CD tracks, the result sounds more natural and less "sterile," as you say, than digital silence ("infinity zero"). In the '50s and '60s, analog tape hiss was more than enough to do the psychoacoustic job.

—Ed.

The Audio Critic:

... You are to be commended for operating one of the best magazines in the audio journalism business....

One problem. In all, or at least most, of your speaker tests you limit your off-axis measurements to no more than 30 degrees. If you read my essay on loudspeaker matters in the April 1995 issue of *Stereo Review,* and study the hemisphere diagram ("Radiation Pattern") and its caption, you can see why I believe it is important for a system to deliver uniform response well beyond 30 degrees. Any decent system can do well out to 30 degrees but only very well-engineered models have uniform response beyond that angle, and those sounds are very audible in normal rooms....

Sincerely,
Howard W. Ferstler
Tallahassee, FL

With regard to your paragraph one: you 're right. With regard to your paragraph two: you 're right.

I wish I had some kind of fancy turntable and tilt mechanism for speaker measurement, so I could conveniently explore the entire front hemisphere of radiation with one microphone. Even so, I think I manage to identify all the good ones and bad ones that cross my path.

—Ed.

The Audio Critic:

...I was reading your Issue No. 23 and came across your comment on page 73: "...not counting independent columnists like Larry Klein and Bob Pease."

Thanks for the vote of confidence. Yes, I am an independent SOB. Yes, I like Mr. Nousaine's work. Yes, I like your work.

Best regards,
R. A. Pease
Electronic Design

Our readers may not know that Bob Pease is a veritable legend in the electronics industry. His praise means a lot.

—Ed.

"An exemplary existing design is *The Audio Critic*, which is remarkable, given its one color (black) printing, and essentially one typeface (Times Roman and its bold/italic variants)."

—From Steve Marston's publication on typography, Feb. 1996. [How about that, tweak-magazine art directors?—Ed.]

Paste This in Your Hat!
(What Every Audiophile Should Know and Never Forget)

By Peter Aczel
Editor and Publisher

If you don't know the ground rules—and you won't find them in the tweako magazines—you'll play a losing game.

All of the following could be proved in court before a jury of degreed professionals—physicists, electrical engineers, acousticians, university professors, researchers in major electronics laboratories.

What is the number one determinant of sound quality in an audio system?

The recording you are playing, without the slightest doubt. The recording microphones, acoustical conditions, and engineering decisions at the recording site introduce much greater sonic variability than any hardware component in a half decent playback system. Buy well-recorded CDs.

What is the number two determinant?

The speaker system, again without the slightest doubt. Even the finest loudspeakers exhibit small irregularities in frequency response, the smaller the better but always audible. Significant differences in f_3 (bass cutoff frequency), efficiency, power handling, distortion, wave launch geometry, and other characteristics result in easily distinguishable sonic signatures from model to model. This is a subject worth studying.

What is next in importance?

The listening room. So important, in fact, that it is hardly distinguishable from the quality of the speaker system itself. It would probably be more accurate to say that the speakers, the room, and the placement of the speakers within the room constitute a single system second in importance only to the program material.

What about the amplifier?

Vastly exaggerated in importance by the audiophile press and high-end audio dealers. In controlled double-blind listening tests, no one has ever (yes, ever!) heard a difference between two amplifiers with high input impedance, low output impedance, flat response, low distortion, and low noise, when operated at precisely matched levels (±0.1 dB) and not clipped. Of course, the larger your room and the less efficient your speakers, the more watts you need to avoid clipping.

What about the preamp, CD player, and other line-level electronics?

As long as they meet the fairly exacting specifications expected these days—and most of them do—they will sound the same, regardless of price. That does not mean, of course, that some are not far superior in measured performance (well below the threshold of audibility) and construction quality.

How important are wires and cables?

No more important than the wiring *inside* your electronics and speakers, over which you have absolutely no control. Speaker cables and interconnects that cost thousands of dollars are a shameless fraud. Radio Shack's reasonably priced top-of-the-line cables are good enough for anyone.

Where do vacuum tubes come in?

Nowhere, unless you are a tweako cultist. There is nothing in audio electronics that cannot be done better with solid-state devices than vacuum tubes. (Maybe—just maybe—the RF stage of an FM tuner is an exception.) Yes, there exists some very nice tube equipment, but the solid-state stuff is better, cheaper, and more reliable. As someone on the Internet said, "tubes are for boobs."

In Your Ear

A man's ambition must be mighty small
To write his name on a toilet wall.
A tweak's ambition is smaller yet
To post a dumb message on the Internet.

The Good Guys in the White Hats and the Bad Guys in the Black Hats: a Guide for the Perplexed

By Peter Aczel
Editor and Publisher

In audio, as in life, there are good guys and bad guys—good and bad manufacturers, designers, dealers, publishers, reviewers, editors, etc. Here you have them conveniently listed for reference.

Who's a good guy? Who's a bad guy? A good guy in the audio world is a practitioner whose efforts, in word or in deed, are aimed at the most accurate sound reproduction possible, at a price commensurate with the means to achieve it. A bad guy in the audio world is a practitioner who has any kind of agenda, overt or covert, contrary to the aforesaid aim of the good guys. It's as simple as that.

Many, if not most, of the names that appear below have been discussed in our pages before, but a consolidated summary of our pantheon and of our demonology appears to be good idea at this point, as we have picked up a large number of new readers in the course of the last few issues. The list is, needless to say, far from complete; there are more good guys and bad guys out there than we could possibly be aware of. The idea here is to answer briefly the incessant questions we get that start with "what do you think of or "how do you rate" or "do you agree with" or "should I believe" and so forth. Of course, those who are familiar with our audio philosophy will readily relate to these lists; the rest of our readers will get the hang of it before they turn the page.

The White Hats

The following audio people have our trust. When you see one of these names, you don't have to proceed with caution. (Please note that our own contributors, such as ***David Rich, Tom Nousaine, David Ranada, Richard Modafferi***, etc., are not incorporated in the list because their White Hat ranking is self-evident—they wouldn't be with us otherwise.)

Robert Adams *(Analog Devices, Inc.)*

Digital audio's voice of authority and silicon jockey supreme. In contrast to those who have only opinions on digital technology, Bob Adams has solutions. Since he designs chips rather than complete audio gear, his solutions affect our audio life only indirectly but nonetheless significantly. On the journalistically and promotionally abused subject of jitter, he is the compass that points true north. Believe him, not the tweako pundits.

John Bau *(Spica)*

Living proof that a typical audiophile can intellectually bootstrap himself to the level of professional engineers. He parted company with the dilettante speaker-builder crowd about 15 years ago and has developed into a thoroughly scientific loudspeaker designer with a valid engineering rationale for every theoretical and practical aspect of his designs. A veritable role model. (Unfortunately, Parasound closed their Spica division after operating it only a couple of years, but I am sure we shall hear from John Bau again.)

Bob Carver *(Sunfire Corporation)*

Possibly the most brilliant audio designer of our time, an inventor rather than just an engineer. His work is nearly always on a level of technological creativity that makes one forgive his P.T. Barnum taste in product naming, advertising, and publicity. His specialty is solving the "impossible" design problem, which he does often. He also happens to be a warmhearted and highly tolerant human being who seldom uses his vast intellectual advantage over not-so-bright critics and adversaries.

Edward Cherry *(Australia)*

Strictly an academic rather than an audio industry person but important to all amplifier designers to this day for helping to straighten out the serious confusion about feedback that existed back in the '70s and early '80s. One of the seminal thinkers in audio electronics.

David Clark *(DLC Design)*

Mr. ABX himself, designer of the original ABX comparator and the earliest apostle of double-blind listening tests at perfectly matched levels. That makes him the tweako camp's Beelzebub, but to my knowledge he has never been proven wrong about what is audible and what isn't. Today his work is mostly in car audio, where ABX comparisons reveal audible differences quite often.

Bob Cordell *(David Sarnoff Laboratories)*

The other great feedback revisionist of the 1980s, together with Professor Cherry (see above). A superbly clear thinker, he is still interested in audio although not part of the industry. His 13-year old prototype MOSFET power amplifier has never been surpassed (nor commercially produced).

Mark Davis *(Dolby Laboratories)*

One of the keenest minds in the industry. He is doing so much highly advanced audio-of-the-future work at Dolby that I am almost embarrassed to remember him mainly for being the first (at least in my experience) to point out that all well-designed electronic signal paths sound the same under controlled listening conditions. That was twenty years ago, when he was still part of the Boston audio mafia, and I didn't believe him. Now I believe all the far more radical things he is saying.

Mike Dzurko *(ACI: Audio Concepts, Inc.)*

The hobbyist manufacturer/marketer who redefined value in loudspeaker systems. Thanks to his excellent taste in sound, his respect for science, and his direct-from-the-factory distribution, ACI speakers have a history of performing like much costlier units sold by dealers. At the moment Mike is working as a schoolteacher again, and the company is in a somewhat austere holding pattern. I trust the situation is temporary because no one is more deserving of audiophile support.

John Eargle *(Delos International, Inc.)*

The Compleat audio expert, a veritable Renaissance man of audio. He has been president of the AES; he has designed loudspeakers for JBL; he has written textbooks and engineering papers on recording techiques, microphones, etc.; he plays the organ and the piano; his credits as a recording engineer go back to the golden age of Mercury and RCA; but today he is best known for making state-of-the-art recordings for Delos. I have never failed to get an erudite, realistic, levelheaded answer from him on any audio question, no matter how controversial.

R. A. (Dick) Greiner *(University of Wisconsin, retired)*

The E.E. conscience of the audio world. We have all learned from him over the years on the subject of amplifiers, wires and cables, polarity—you name it, the list is endless. He makes life a little simpler and easier for those of us who trust science because he is a great explainer. To tweako cultists he is a nemesis because his calm professorial logic devastates their agenda.

David Hall *(Velodyne Acoustics, Inc.)*

The emperor of subwoofers and defender of the faith (not shared by all practitioners) in low-distortion loudspeaker design. His motional-feedback subwoofer design of 1989 signaled the beginning of a new era in bass reproduction. He is very much a hands-on engineer, and his latest stuff is still ahead of the competition.

Ken Kantor *(Now Hear This, Inc.)*

Another original whose unconventional ideas on sound reproduction, more specifically on loudspeaker design, must be taken seriously. He is so smart that he has gained entree into, and the confidence of, tweako circles without being a tweak himself. Neat trick.

D. B. (Don) Keele, Jr. *(Audio magazine)*

The most honest, thorough, knowledgeable, and commonsensical of loudspeaker reviewers. (Present company excepted? Hell, no.) He is responsible for the highly reliable and accurate nearfield method we all use now to measure woofers. Unfortunately, just because he is a total objectivist, it does not follow that *Audio* shuns tweako reviewers of questionable credibility.

Siegfried Linkwitz *(Audio Artistry)*

One of the truly serious thinkers on the subject of loudspeaker design, with impeccable academic and professional credentials. His widely quoted work on crossover networks provided the antidote to the simplistic first-order cult. His current work on large speaker systems shows considerable originality.

Stanley Lipshitz *(University of Waterloo)*

Arguably the keenest intellect in the audio community. He is not associated with any specific audio product but has mathematically analyzed just about every important audio design problem and written a paper about it (usually with fellow savant John Vanderkooy—see below). If we don't understand something about a new technology, we ask Stanley. He knows. As David Clark (see above) once said, "The audio world doesn't deserve Stanley." That's probably true, but as Clint Eastwood said in *Unforgiven,* "Deserve's got nothing to do with it." We have him, and he is indispensable.

E. Brad Meyer *(The Boston Audio Society, CompuServe)*

One audio journalist who makes a serious effort to be objective. His tests and his writings evidence both technical knowledge and intellectual honesty. If he has an audio-political agenda, I am not aware of it. His well-documented article on the CD vs. vinyl controversy in the January 1996 issue of *Stereo Review* is a case in point. He is also a recording engineer and producer.

Ed Mutka *(B&K Components, Ltd.)*

A circuit designer after our own heart. He does pretty much everything right, even though he must operate within budget constraints. What's more, he is not afraid to express strong opinions about the right and the wrong way to design audio equipment. His boss, ***John Beyer***, also deserves full credit for the intelligent guidelines that allow Ed to do his thing.

John Ötvös *(Waveform)*

Not an engineer but an audio perfectionist advised by some of the best engineering brains. Almost painfully honest and uncompromising, he takes the high road of scientific loudspeaker design without heed to trendy directions or commercial pressures. Possibly the most selfless, idealistic man of audio.

Ken Pohlmann *(University of Miami, Stereo Review)*

The straight talker of the digital domain. Read his textbooks if you want to be genuinely savvy on digital matters; read his magazine articles for general insights; in either case you will be totally safe from the digital drivel that permeates so much of the audiophile press. His academic specialty is actually "music engineering."

Chris Russell *(Bryston Ltd.)*

The incorruptible amplifier designer. His designs combine engineering elegance with moral rectitude (meaning the simplest solutions that will yield maximum performance, no expense spared where it counts, not a penny for tweako fetishes). That goes double, with little bells on it, since ***Stuart Taylor*** (ST) became his engineering associate.

Jim Thiel *(Thiel Loudspeakers)*

The high priest of the doctrine of coherence through first-order crossovers in loudspeaker systems. I do not even agree with his doctrine but nonetheless admire him for his engineering talent and uncompromising integrity. The man's devotion to scientific design and quality construction cannot be questioned. On top of it he is a true gentleman.

Floyd Toole *(Harman International Company)*

The man who codified the controlled subjective testing of loudspeakers and the listening/measuring correlation. His scholarly work at Canada's National Research Council laboratories was so basic and groundbreaking that Sidney Harman, whose appetite for audio talent is insatiable, just had to have him and his associate ***Sean Olive.*** At this point I'm still waiting for JBL, Infinity, Citation, and/or other speakers from the Harman group that fully reflect his input. They ought to be good.

John Vanderkooy *(University of Waterloo)*

Physicist partner of Stanley Lipshitz (see above). They are team; their names appear as coauthors on AES papers; however, John maintains a much lower profile vis-à-vis the audiophile world, so that only a professional inner circle is regularly exposed to his awesome intellect and urbane charm. I have seen him in action as a B.S. detector and audio-issue clarifier, and I am a believer!

Kevin Voecks *(Harman International Company)*

Long associated with Snell Acoustics, now embarked on Sidney Harman's latest high-end project, Revel loudspeakers. I identify with his evolution from wide-eyed audiophilia to hard-nosed scientific objectivism because it paralleled mine. His last few Snell speaker systems have been nothing short of world-class.

Max Wilcox *(independent producer/recordist)*

The name that guarantees integrity in the interface of music and sound. As a fine musician, Max is able to steer a recording session to the highest performance level obtainable with the given talent, and as a recordist of unfailingly good taste he will tape the performance in utterly natural, accurate, unexaggerated sound. His recordings (on RCA, Elektra/Nonesuch, MusicMasters, Teldec, DGG, etc.) seem to wear better on repeated listening than the work of the hotshot, razzle-dazzle engineers.

Sao Zaw Win *(Win Research Group, Inc.)*

Who says there is a conflict between aesthetics and science in audio? This Burmese-American technologist excels in both. His creations are invariably beautiful (in looks and in sound) and scientific (in concept and in execution). If all practitioners were like him, the audio world would be Camelot. Among other things, he is probably the number one authority on materials science as it relates to transducer design. Paradoxically, he seems to be irresistibly attractive to the tweako community, whose pundits are constantly courting him and whom he constantly rejects! I have written so much about him in the past that this much should be sufficient here.

The Black Hats

This is a much more difficult list to put together because, on the one hand, I have no desire to pull any punches and, on the other hand, I want to avoid being grossly unfair. Needless to say, the commentaries below are simply opinions, my own opinions, although in most instances they are shared by the audio people I respect.

Larry Archibald *(Stereophile)*

The *arch-* in his name signifies archenemy—the principal, most powerful enemy of scientific discipline in equipment evaluation and of accountability in audio journalism. No, there's nothing fiendish about him as a person; he is a nice, intelligent guy to have a drink with; but he is a total opportunist as a publisher. His magazines tell you what he believes you want to hear, because that's where the money is, not what you ought to know, namely the unvarnished realities of the subject matter. I am convinced he knows what those realities are; he just doesn't think they are moneymakers.

John Atkinson *(Stereophile)*

Highly intelligent, extremely competent, transparently insincere. I don't know when the hypocrisy started; maybe in his earliest days at *Hi-Fi News & Record Review* in England he actually believed the tweako B.S. he now redacts and asseverates in Santa Fe; but I refuse to believe that he still believes it. He has been exposed to too much overwhelming scientific evidence to the contrary and he just can't be that dense. (See also Issue No. 22, p. 10.) At this point he mechanically reiterates the party line and comes up with progressively more tortured sophistries to bolster it. Why? Because his job at Larry's place requires it, and it's a good job. The trouble is, he has too many readers who still take all that rubbish at face value.

Bruce Brisson *(MIT: Music Interface Technologies)*

Grand master of the specious technical argument promoting insanely expensive tweako cable. I find his approach particularly sleazy because he uses intimidating buzzwords and icons of science and mathematics to lead the audiophile to false conclusions—he hopes you won't grasp the total irrelevance of his highfalutin general analysis to the transfer of audio frequencies over short distances in a domestic audio system. He knows exactly what he is doing, and that makes him despicable.

William Conrad & Lewis Johnson *(Conrad-Johnson)*

The duo chiefly responsible for, or at least heavily contributory to, the cult of formatted vacuum-tube sound, achieved with deliberately high output impedance (i.e., low damping factor) and lots of second harmonic distortion. In his famous/notorious "t-mod" soundalike experiments of the mid-1980s, Bob Carver had to screw up a perfectly neutral solid-state signal path to make it sound exactly like a Conrad-Johnson tube job. Not much has changed since.

Anthony Cordesman *(Audio, formerly TAS)*

I have had my say about him in "Hip Boots" (Issue No. 20, pages 62-63), but he is a major Black Hat and needs to be be listed here. He illustrates the intellectual tragedy of a tweako audio culture that can ensnare a highly intelligent and widely respected expert in another discipline (military analysis, national security) to the point where he writes crashing stupidities about audio equipment. An unforgivable shame.

Michael Fremer *(Stereophile, formerly TAS)*

Possibly the most unattractive individual in the American audio community. In his writings and in his personal contacts, he is vulgar, abusive, bigoted, and intellectually dishonest. A real charmer. His favorite cause is the superiority of vinyl to CD, an argument he pursues to the limits of absurdity and animosity, making a total jackass of himself in the process. A perfect example of the excesses engendered by tweako cultism and a highly suitable addition to the *Stereophile* stable.

Corey Greenberg *(Audio, formerly Home Theater)*

Talk cool, think tweak—that seems to be his creed as an audio journalist. He obviously believes that semi-educated, unaccountable subjective reviewing is more credible if the style is late-1960s *Rolling Stone* gonzo. The trouble is that in the mid-1990s such a style is no longer cool; as Lou Reed (who really *is* cool) would say, stick a fork in it and turn it over, it's done. Occasionally I discern a faint glimmer of technical insight in Corey's undisciplined opinionfests, suggesting that he might have the potential to be a good reviewer if only his intellectual environment were totally different. It remains to be seen whether his mid-1996 switch to *Audio* magazine constitutes a corrective environmental change. Time will tell.

Dennis Had *(Cary Audio Design, Inc.)*

The single-ended triode amplifier man. That concept is such an outrageous piece of stupidity in the context of mid-1990s technology that any technical debate about it is merely embarrassing. Meanwhile such highly innovative and sophisticated amplifier designs as those of Bob Cordell and Mark Alexander (see Issue No. 20, p. 22 and p. 25) go without commercial implementation. Hopeless marketplace...

Robert Harley *(Stereophile)*

The most influential and, at the same time, least qualified writer on digital audio (among other subjects). He got his job at *Stereophile* by winning an essay contest, for crying out loud! There seems to be no evidence that anyone asked him for his academic/technical credentials. His blunders on the test bench and on the printed page are the laughingstock of degreed engineers and academics; he has been skewered and punctured both in print and face-to-face so many times that he resembles a sieve; but he keeps plowing right ahead with his flawed tests and reviews—and his publisher (Archibald) and editor (Atkinson) let him! He appears to believe every half-baked tweako cliche ever put forth, and in his case I am almost certain he is sincere. (The more the pity—but a deluded Black Hat is still a bad guy.)

William Z. Johnson *(Audio Research)*

Take all the vacuum-tube B.S. you have ever been exposed to, trace it back—who heard it from whom, who read it where, who said it first, etc.—and I think you'll end up with Bill Johnson as Genesis 1:1. (I don't mean the electronic theory, which goes back to 1907; I mean the tweako amplifier bandwagon.) It is also my impression that he was the very first manufacturer to realize, in the late 1960s, that you can charge a hell of a lot more for a piece of audio equipment than its true value in terms of parts, labor, and R & D. That insight started a whole industry based on image and style rather than substance. His black hat is therefore size XL.

Ray Kimber *(Kimber Kable)*

A pair of 4-foot speaker cables for $15,000? Yes, that's the man. He is totally shameless. What would go better with those spiffy cables than a nice black Stetson? (I must add that there's nothing wrong with the more or less reasonably priced cables he also makes.)

Noel Lee *(Monster Cable Products, Inc.)*

Father of the tweako cable industry. He was the first to realize that you can make lots of money in high-end audio without a factory, without engineers, without a technological innovation, without any talent other than salesmanship—in other words, without significant overhead. All you need is a marketplace of gullible audiophiles, a steady stream of B.S. promising amazing sonic improvements, and an account with an established maker of wires and cables (Alpha, Belden, Canare, Mogami, or whatever). You specify various tweako configurations, your supplier delivers the cable with your brand name on the insulation, you set an astronomical price, you give the dealer 50-plus points, and—voila!—you are a legend of the audio salons. Noel Lee adds a sophisticated credibility device to the formula: he also sells perfectly normal, conventional, high-quality wire, cable, and connecting hardware to the trade, at a fair price. His black hat is covered by a white hat on the outside. Smart boy, that Noel.

David Manley *(Vacuum Tube Logic)*

One of the chief technical apologists of the tweako vacuum-tube cult. He wrote a book on the subject and he does, or at least used to do, some heavy proselytizing at the trade shows. Nice shtick; too bad he is dead wrong.

Arnie Nudell *(Genesis Technologies, Inc., Eosone)*

Ask him about making a silk purse out of a sow's ear. He has perfected the technique. The "heart" of the Genesis I, his $90,000 flagship speaker system—i.e., the driver that determines its basic sound—is the five-foot ribbon from the old Carver "Amazing Loudspeaker." It's an excellent transducer; it sounds great; but it's ingeniously fabricated out of cheap parts and was originally developed for a speaker system which, in its most expensive form, sold at about 3% of the Genesis I's price. I estimate that Arnie gets the ribbon from Carver Corporation at an OEM price of maybe $100 each, certainly well under $200. Yes, there are many other drivers in a Genesis I (that's one of its problems, actually) and very handsome cabinetry, but $90,000? Designed around the Carver ribbon? If at least they admitted it—but when I first pointed it out more than two years ago, they freaked. And that's not the only skeleton in Arnie's engineering closet...

Harry Pearson *(The Absolute Sound)*

After my full-length portrayal of the man in the last issue (No. 23, p. 72), I don't want to belabor the point here. He is *the* most grotesque embodiment of half-assed, tweako cultism in audio. At this point in his dismal decline I'm not even sure he is playing with a full deck. Next!

George Tice *(Tice Audio Products, Inc.)*

The original power-conditioner flimflam man. Not to mention the clock you plug into your wall for a miraculous improvement in sound. One must admire the unmitigated gall of the man while dismissing his B.S.

David Wilson *(Wilson Audio Specialties, Inc.)*

An unlikely Black Hat because he is a nice, intelligent, highly civilized gentleman and a super recordist. But he also happens to be the godfather of the mega-priced speaker racket. The Wilson WAMM system costs $147,000 and has nothing in it that justifies even a fraction of that price. The 40-odd rich audiophiles who bought it over the years ended up with truly superior sound, such as you can get with a (say) $15,000 speaker system, and thus remained perfectly happy because the $132,000 overcharge (all right, a few thousand less at earlier prices) didn't mean a thing to them financially but boosted their audiophile egos tremendously. That doesn't make David Wilson a White Hat, however, nor does it put clothes on the emperor.

* * *

I must add that ***nearly all of the very high-end dealers from coast to coast*** are Black Hats because they are stuck with the tweako party line. They have to tell you that the upper midrange is more liquid with single-ended triodes or that the silver cable has better rhythm and pace, otherwise they can't sell the stuff. I hesitate to single out the stores that have personally nauseated me, as there may be others unknown to me that are just as bad or worse.

Patriotism is the last refuge of a scoundrel.

—SAMUEL JOHNSON (April 7, 1775)

"Musicality" is the last refuge of a tweak.

—PETER ACZEL, *The Audio Critic* (in a long-ago issue)

I regard "double-blind comparative listening tests" as the last refuge of the agenda-driven scoundrel.

—JOHN ATKINSON, *Stereophile* (December 1996, page 23)

[Draw your own conclusions. So Clark, Toole, Olive, Lipshitz, Bech, Nousaine, &c, are scoundrels?—Ed.]

Consumer and Designer Prejudices in High-End Audio: A New Way to Examine Them

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

We look at high-end audio circuitry not from the testing point of view but as an engineering discipline—and find no consistency.

Editor's Note: *This article is based on two engineering papers delivered by David Rich in 1995 before two different professional societies. The first was an advanced tutorial at the DSPx '95 conference in San Jose, CA, addressing the question: "Are there any design considerations in audio that go beyond standard measurements?" The second, coauthored by your Editor, was presented at the 99th Convention of the Audio Engineering Society (October 1995, New York) under the title "Topological Analysis of Consumer Audio Electronics: Another Approach to Show that Modern Audio Electronics Are Acoustically Transparent" (Preprint 4053). What follows here is a digest of these two professional papers, revised and edited for the audiophile consumer. Technical details left out here can be found in the AES paper. We ask our readers' indulgence for the few audiophile commonplaces that have been left in; these are not always obvious to professional engineers who don't read the consumer magazines. The professional perspective, on the other hand, is one of the benefits the audiophile can derive from this material.*

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Degreed electrical engineers tend to regard the design of audio electronics to be trivial compared to other design challenges of consumer electronics, yet it is more complex than appearances would suggest. Some of the design challenges may not relate to measured performance but instead to consumer expectations and beliefs.

For example, it is not possible to sell an audio product with a sharp-cutoff analog reconstruction filter. Consumers have been told by dealers, sales literature, and audiophile magazines to believe that the asymmetric ringing on square waves associated with such a filter has a detrimental effect on the audible quality. As a result, interpolating digital FIR filters placed before the DAC are used for the reconstruction function. This digital approach results in symmetrical ringing of the square waves, which consumers have been informed is insignificant. In this example, we see the first occurrence here of those troubling words that keep coming up in audio design: *believe* and *audible.*

Controlled vs. open-loop listening tests.

The problem with audio design is that the performance is assessed by listening to the unit. The subjective nature of the assessment of audio components makes it possible to make outlandish claims for the audible characteristics of a piece of audio equipment. Outlandish claims are more easily made for comparisons involving hearing than those involving vision. When people claim that they see something different, they can physically point to the difference to substantiate that the difference exists. This is not possible when someone claims to hear something that another listener cannot [Eargle 1995]. Audio dealers constantly invoke the phrase "Can't you hear that?" as they attempt to close a sale. Fortunately, controlled listening-test methodology can be used to eliminate much of the uncertainty of making a subjective evaluation.

Unfortunately, comparisons of audio equipment at the consumer level are never done with double-blind tests. The "buff books" that consumers of audio equipment read are also not using controlled subjective listening testing (except *The Audio Critic).* They instead use unreliable open-loop testing. The brands of the equipment are known to the reviewer, and levels are not carefully matched. As a result, elaborate descriptions of the sound quality of the equipment appear in print. As would be expected, given that the testing methodology is so sloppy, little correlation exists in the descriptions of the sound of a component from reviewer to reviewer. Indeed, even those who support the open-loop listening method can only cite loudspeaker reviews as consistent in describing the sound of a given product [Harley 1991]. The only thing that does appear to correlate in the reviews is that more expensive equipment is felt to sound better than less expensive equipment.

Why should the magazines want to perpetuate such sloppy testing? The answer has two parts: (1) The high-end audio industry depends on convincing consumers that large differences in components exist and that expensive components are clearly better than cheaper ones. Buff books expounding this philosophy have increased the number of their advertising pages from both dealers and manufacturers. (2) Controlled listening tests may be difficult to set up, and doing multiple listening trials represents real work. Most of the writers in consumer magazines are having too much fun doing open-loop listening tests to want to change to a methodology that requires real work.

Controlled listening tests have consistently shown that electrical components will be audibly indistinguishable if the have: (1) flat frequency response, (2) noise and distortion levels below audible thresholds, (3) high input impedance and low output impedance [D. Clark 1982]. So how can the audio magazines justify the differences they report, when the measurements they make on the equipment show that well-designed audio electronics have virtually flat frequency response and noise plus distortion 80 dB or more below the fundamental test signal? The answer is that they use pseudoscience. Consider the following quote from *Stereophile* magazine [Harley 1990], discussing modifications to CDs that make no technical sense.

"I see CD tweaks as a Rosetta Stone to an audio engineering establishment that dismisses the possibility that freezing a CD, or painting it black, or putting green paint around the edge, or making it from a different material, could affect its sound. Because these treatments are considered the epitome of audiophile lunacy *and* because they are readily audible, some measurement-oriented scientists may, if they listen for themselves, realize that audiophiles are not always the demented mystics they are often accused of being."

This idea that undiscovered phenomena are responsible for the discrepancy between measurement and audible observations is commonly expressed in the audiophile press. It should be noted that none of the items discussed by Harley have ever been shown to be audible in controlled listening tests.

Often the semitechnical (to use a kind word) audio writer may try to claim that a measurement that can be made only inside the equipment is responsible for the sonic performance. For example, it has been reported—once again in *Stereophile* [Harley 1994], but there are also other examples—that CDs which have more eye pattern closure at the photo detector output of the CD player have poorer sound. The fact that the designers of the CD playback system designed the system to be insensitive to eye pattern closure, as a result of reclocking of the data by a stable crystal reference, is never discussed. Because the signal is reclocked, it is impossible to see any effect of the closure of the eye pattern at the CD player's output jacks. Other examples of this kind of untutored reporting are discussed in [Aczel 15, 17].

Disinformation for the innocent consumer.

It is important to keep in mind that "technobabble" concerning audio equipment is not confined to the buff books but has spread to such highly respected business publications as *Forbes* and *Business Week* [Aczel 16, 22], which have both published articles claiming that tube amplifiers produce sound quality superior to that of solid-state units. As a result, consumers who are not directly involved in the hobby may still be led to believe many of the strange notions of the audiophiles. Just as in the audiophile press, these articles quote no electrical engineers to give a plausible explanation of why a tube amplifier might sound different, such as the high distortion and high output impedance measured in typical tube amplifiers.

Improperly conducted subjective testing methodologies have given rise to an entire industry that produces what can only be described as consumer rip-offs. These products cannot possibly affect the sound quality of a system. They exist because it is possible to convince the purchaser through techniques of salesmanship that a sonic change has occurred when no real change has occurred. Examples include: (1) a $400 LED clock claimed to improve the sound of a system when it is plugged into the same power line as the stereo, (2) brick-shaped objects of wood filled with ferrous metal inside that are claimed to improve the sound quality when placed on a stereo component, (3) a device which generates signals to be sent through an interconnect or speaker cable for the purpose of "burning the cable in" to obtain better sound, and (4) the CD "tweaks" mentioned by Robert Harley in the quote above.

Although it is possible to sell consumers the above products despite the fact that they do nothing, it becomes even easier to sell products if they do change the sound, and it turns out that some audiophile products can yield a definite sonic change which is observable in controlled listening tests. Audiophiles have a tendency to assume that if a product sounds different it must be better, but in most cases the differences can be attributed to the introduction of frequency response errors or the addition of distortion. The single-ended low-power class A triode amplifier craze is an example of this.

One easy way to introduce frequency response errors is to increase the output impedance at the amplifier-loudspeaker interface by inserting a series inductance or resistance. This is exactly what is done by very expensive speaker cable. These cables, which can cost over $1000, do nothing but add a series inductance and series resistance. Expensive power amplifiers often have very high output impedance, which will affect the system transfer response in a similar manner. Cable manufacturers often take advantage of the high output impedance of such power amplifiers to further modify the transfer function. This is done by designing the speaker cable to have significant parallel capacitance. Frequency response plots showing these effects can be seen in Issue No. 16 [Aczel 16]. Significant errors in the RIAA equalization curve of a phono preamp are often present in designs preferred by audiophiles. Similar things can be done in the digital domain. For example, coefficients in a digital filter have been derived in the time domain, using spline functions, by manufacturers such as Wadia [Moses 1987]. It is claimed that better sound results from this. While such a claim is highly debatable [Lipshitz 1991], it is not debatable that this technique results in the frequency response being down by 3 dB at 20 kHz, and that response change is audible.

Nonlinear distortion mechanisms are often present in designs preferred by audiophiles. One recent fashion that has proved popular among audiophiles is a preference for single-ended amplifiers. Such amplifiers have significantly higher levels of distortion than push-pull designs. Distortion well over 1% is observable in these amplifiers. Audiophiles have often shown a preference for amplifiers with little or no feedback because they believe that feedback somehow makes the sound quality worse. In the real world, amplifiers using little or no feedback will exhibit increased distortion, especially at lower frequencies. Again distortion of 1% or more has been observed in low-feedback designs, although careful use of local feedback and other innovative techniques can bring the distortion below the level of audibility.

The designer's dilemma.

Given the confusing trends and frequent irrationalities of the consumer audio market, how is a designer to respond when assigned the task of designing consumer audio components? Three approaches can be taken by the professional engineer: (1) Modify the frequency response of a product, or add distortion to the product, so that it will have a different sound characteristic and thus achieve a competitive advantage; (2) ignore any considerations of sonic issues and design to achieve a design with flat response, low distortion, low noise, high input impedance, and low output impedance; (3) design the circuit so that it is judged to be transparent under controlled listening tests, and let the sales and marketing department justify that the design is superior.

A key question for the designer to answer is whether or not approaches 2 and 3 will lead to the same results. If they do not, then what are the design parameters required to achieve sonic transparency? This is the question that we seek to answer here. Our own experiences have shown that when a piece of stereo equipment passes the traditional set of measurements (distortion, noise, frequency response, input impedance, output impedance), then it will not show any acoustic signature in a controlled listening test. But the marketing department must be aware that audio components that achieve excellent specifications at relatively low price points are often not the market leaders. Indeed, one of the most remarkable products we have tested, the Boulder 500AE stereo power amplifier, based on a topology discussed in [Jensen 1980], achieved distortion levels as low as a perfect 16-bit converter at full power across the audio band, but it has not been a major success. We can see the problem in the reaction to the excellent measured performance of this amplifier by one reviewer, who wrote the following [Harley 1992]:

"...the 500AE was designed on paper, rather than in an iterative listen/design/listen process. ...I believe the Boulder 500AE to be good, but not extraordinary..."

Contrast the above reaction, to an amplifier that measures really well, with the following quote, from a review of a Jadis tube amplifier [King 1986]:

"So why do these amplifiers sound so good? Now my job gets tough. The bottom line: I don't really know. The only really good measurement is of its harmonic structure. Output impedance and amount of harmonic and IM distortion are OK, but, the gross slewing or reduced high frequency power output is bad. The truth is that the measurements most of us make are not very relevant to the sound of circuits and I've spent a good part of my career looking for ones that do with little success so far."

In no other field would this statement be taken seriously. It appears very likely that this amplifier was perceived as sounding good because it falls into category 1.

It should be noted that one possible approach for the designer is to work in category 1 and then disclose what he is doing. This is the approach that has been used by Bob Carver. He modified the transfer characteristics and output impedance of a low-cost solid-state power amplifier to match a very expensive tube amplifier that the audiophile community thought had the best sound quality. In double-blind listening tests he then demonstrated that the two amplifiers sounded identical. Unfortunately, the consumer audio press and the distribution channels reacted very negatively to Bob Carver's approach. Their reasoning was obviously that if Mr. Carver's approach were given legitimacy, it would no longer be possible to sell the expensive tube amplifier and other very expensive audio components. Bad publicity from the audiophile press reduced the sales of the Carver amplifier dramatically. Understanding how a given engineering approach will be accepted by the press and the public is an important issue for the marketing department, but should not be the concern of the designer.

If the designer wants to work in category 2, he needs to know what are the limits on the electrical specifications that must be achieved for a unit to be audibly transparent [Olive 1995]. There is a wide range of answers, depending on which investigator's result you are reading. Signal-to-noise ratio clearly needs to be better than 98 dB if one wants to pass a 16-bit digitally encoded signal without degradation. It has been shown that in some cases a dynamic range greater than 98 dB may be required to capture the full dynamic range of music [Fielder 1989]. Because the ear is more sensitive in some frequency bands, noise shaping can be used to increase the subjective dynamic range of a 16-bit system. If noise shaping is used, the signal-to-noise ratios of the D/A and analog sections may need to be in the range of 110 dB [Benjamin 1993] if the noise in these sections is flat over the audio band. With respect to distortion, most studies have shown that 0.1% distortion is inaudible even when using test tones. Other studies have shown that distortion on the order of 2% is not audible on music signals [D. Clark 1982].

The ear appears to be very sensitive to level differences, and overall level mismatches larger than 0.1 dB may be perceived. Localized deviations of frequency response may require larger deviations to be audible. The upper frequency response limit is typically determined by the sampling rate of the signal if it has been, or will be, digitized. At the low end, the loudspeaker usually rolls off well before electronic components. One consideration that causes some designers to place the low-frequency rolloff at a very low frequency is phase errors at 20 Hz. It has been shown that the ear is slightly sensitive to phase errors in the bass spectrum [Fincham 1985].

Input and output impedance specifications are principally determined by frequency-response considerations. In most cases these errors are most likely to occur at the amplifier-speaker interface, but it is possible to observe high-frequency rolloff at low-level signal interfaces [Aczel 17]. This is almost always due to a component with a very high output impedance (3 to 50 kΩ) driving a cable with very high capacitance (1 nF).

A new way to evaluate design differences.

Is it possible that a design accomplished with the approach in category 2 will not lead to a transparent audio component? One way to answer this question is to look at already existing designs that are claimed to have excellent sonic qualities on the basis of open-loop listening tests. By examining the circuit topologies, we should be able to determine if some new design approach is being used that would not have been used if the design had been developed simply to achieve good results in traditional bench measurements. Such examinations [Rich 15, 18, 20] have shown that these designs do not show any particularly unique circuit-design topologies. Furthermore, little commonality can be seen among the designs' topologies.

It is a reasonable assumption that a superior audio-circuit concept would slowly but surely recruit a much larger following than an inferior one, so that eventually there would be some kind of consensus among practitioners and a discernible convergence toward the superior topology as new designs emerge. Total randomness in the choice of topology would indicate that no single approach is clearly superior to any other, in much the same way as total randomness in the results of a double-blind ABX test indicates that there is no clearly audible difference between A and B. It's basically the same statistical criterion. Thus, if a designer had discovered a unique topology that sounded better but did not measure better, we would expect that topology to spread to other companies through "reverse engineering" of the product with the superior sound. There is no evidence whatsoever that this is actually happening.

Although nothing especially unique can be found in the design of audio components as distinct from other electronic products, a number of circuit-design techniques can be seen in high-end audio equipment that are distinct from common design practice. As stated previously, we have found that none of them yielded a product with superior sound quality, but it is still interesting to look at them to see if they can reveal any clues regarding the thought processes of the designers of the equipment. Note that the summary below has been generated on the basis of many different pieces of audio equipment, with no single one of them using all or even most of the techniques outlined.

1. Discrete circuit design. High-end audio designers tend to shun integrated op-amps. This gives them the added flexibility to design circuits such as are discussed below, which they would not have designed with a single monolithic device. The principal advantages of discrete design from a measurement point of view are reduced noise levels and increased output drive capability. The downside is that the circuit will be slowed down because of larger parasitics, and it may cost more. The former problem is not a significant concern at audio frequencies.

Of course, many high-end products still have integrated operational amplifiers in the signal path. The op-amps may often precede or follow the more exotic discrete circuitry favored by the high-end designers. Since the op-amps have none of the design features that these designers believe are required to prevent sonic degradation, such practice appears very strange. For example, high-end designers who hold that an amplifier must use very low levels of feedback to prevent sonic degradation may use op-amps that have high feedback levels at low frequencies in the signal path [Rich 21]. Despite their deviation from the "politically correct" techniques common to high-end design, electronics with integrated operational amplifiers in the signal chain often receive very favorable reviews. Go figure.

2. Extensive use of FETs. Designers' typical explanation for this is that FETs perform more like tubes. A more scientific explanation involves the fact that these devices increase the input stage's dynamic range (see 6. below) and that they are more robust into short circuits (see 3. below). Some IC manufacturers often encourage the use of BiFET op-amps for better sound quality using the explanation that they have a distortion characteristic more like tubes [Burr-Brown 1992]. This begs the question that, if the distortion numbers are very low, why does it matter what the characteristics of the distortion are.

***3. Class A output** stages.* Audiophile folklore has always stated that class A amplifiers sound better. This can be carried as far as biasing the output stage of a power amplifier into class A. Clearly, crossover distortion is possible in class AB stages, but this can be minimized [Sandstr0m 1983] in good designs. If the designer is using an op-amp, he may put a load resistor from the output to the negative supply rail to cause a large dc current to flow. This dc current forces the *npn* output transistor on for the full swing of the output, yielding class A operation [Jung 1986]. One design we have examined got this backwards by placing the resistor to the positive supply rail, forcing the slow *pnp* transistor on instead. Despite this the amplifier has received good reviews. (The old football-team adage "what the ref don't see don't bother the ref appears to apply to high-end audio reviewers as well.)

Some power amplifiers are stated to be class A by manufacturers so that audiophiles will think they "sound good," even when they are in reality class A/B amplifiers with high quiescent currents levels. Other amplifiers use dynamic biasing circuits which keep the output device that is not driving the load biased to a small constant quiescent current. This technically fits the definition of a class A amplifier, but since the output devices still experience wide variations in current flow, the problems identified by Sandstr0m still apply. This dynamic biasing approach has been adopted in nonaudio applications that must run at low power-supply voltages [Sakurai and Ismail 1995]. The designers of these dynamically biased output stages still refer to them as class A/B and never class A. Audiophiles are not aware of such distinctions and in open-loop listening tests they find amplifiers labeled class A to have excellent sonic qualities in comparison with class A/B amplifiers—even if the so-called class A amplifier passes into class A/B above a certain signal level or if the amplifier labeled class A uses dynamic biasing.

Other design trends in output stages can also be seen. In low-level signal stages, current limiting circuits are not often used. Resistors in series with the output provide the current limiting. In power amplifiers, current limiters may still not be used, with designers relying on rail fuses to protect the amplifier. If current limiting is used, it will not be the simple one-transistor foldback design. It can be shown, using some novel test procedures, that an improperly designed current protection circuit can activate prematurely into real loudspeaker loads [Otala 1987], [Baxandall 1988]; thus, relatively sophisticated protection circuits are required. The effect of carefully designed current-protection circuits can be measured by checking to see if the voltage output is reduced when driving reactive loads. Often these tests must be done on a dynamic basis because the amplifier may have inadequate heat sinks for steady-state testing [Rich 20]. It should be noted that many amplifiers said to "sound good" in open-loop listening tests have protection circuitry that operates poorly.

4. Low levels of global feedback. One of the parameters often identified as important is the amount of global feedback that may be applied in the circuit. This concept was first identified in [Otala 1970], although Otala's analysis has not been accepted by some peer reviewers [Cherry 1982], [Cordell 1980], [Cordell 1983]. In its simplest and most common form, the open-loop gain is made constant up to the 10 kHz to 50 kHz range. This is accomplished by resistively loading the voltage-gain stages in the amplifier. This approach increases distortion at low frequencies, since the return-loop gain is held constant across the frequency band. Gain stages often have large amounts of emitter or source degeneration in an attempt to linearize the gain stages with local feedback, so that global feedback rates can be reduced. It is well known from feedback theory that multiple small feedback loops will not be as effective as one global loop, so the local feedback approach cannot be justified on pure engineering grounds [DiStefano 1990]. Some designs may have no global feedback or very small amounts of feedback (6 dB or less). In these cases designers must move beyond local feedback and use more exotic methods of error cancellation [Cordell 1984].

5. Fully complementary circuits. These circuits are designed to be fully complementary from the input stage onward. This technique may be useful in reducing distortion in amplifiers that are run at low feedback levels because even-order harmonics are canceled. One reason this is helpful in low-feedback designs is that, when large amounts of local feedback are used in the second gain stage, the voltage swing at the input of the stage must be larger, since the stage's gain is reduced. As a result, the first gain stage's output swings are higher, and this stage can now contribute significant distortion. The downside of a fully complementary amplifier is increased noise, dc offset, and decreased CMRR.. Another clear disadvantage is increased parts count.

While a fully complementary amplifier may be overly complex and may not always yield the best performance, one aspect of its design can lower distortion over the standard topology. This aspect of the design is the push-pull second stage. This is especially true in power amplifiers, where a large voltage swing occurs at the output of the second gain stage. Achieving a push-pull second gain stage while retaining the high low-frequency gain in the first stage is not a trivial design problem. There exists a published circuit which does this [Cordell 1984]. This circuit, combined with cascodes in the second stage and a buffer stage between the first and second stages, results in an amplifier with remarkably low distortion levels. Despite its many advantages, the amplifier has never found a commercial realization—perhaps because it measures too well to "sound good" to the indoctrinated open-loop listener.

6. Input stages with very wide open-loop linear range [Jung 1987]. A designer who searches the literature will see the term *transient intermodulation distortion.* In brief, this effect occurs when an amplifier slew limits. Early work suggested that transient intermodulation distortion could be eliminated only if small levels of global feedback were used [Otala 1970]. Later work showed that the effect could be eliminated if the input stage linearity was made large enough so that under worst-case conditions the summing junction is never moved outside the linear range of the input stage [Leach 1981]. Designers will use FET devices and/or degenerate the gain device in the front-end differential pair to achieve the wide linear range.

Some researchers have suggested that this requirement results in significant overdesign and that the transient intermodulation effect cannot occur with band-limited music signals [Cherry 1986]. This explains why bipolar op-amps with no degeneration of the differential stages are acoustically transparent in controlled listening tests. Also note that sophisticated tests are not required to test for transient intermodulation distortion. If an amplifier has low levels of THD at 20 kHz on full voltage swings, it is free of transient intermodulation distortion. If an inband test is required, then some unusual three-tone intermodulation tests can be used [Borbely 1989]. Examining the input stages of amplifiers said to "sound good" will show that the amount of degeneration varies significantly from no degeneration on a bipolar stage to orders of magnitude beyond the emitter (or source) resistance of the active device.

7. Radically overdesigned power supplies. It is not uncommon to see power transformers and rectifiers much larger than required to drive the power supplies. Multiple stages of regulation are often used [Jung 1995]. Sometimes this is carried to the point where each op-amp has its own local regulator. Discrete regulators are often used in places where a cheaper monolithic device would have been sufficient. Designers will explain that the regulator has to be "fast" to improve PSRR. Exactly why simple bypass caps are not as good is never made clear.

Again, no consistent design practice is observable, since inexpensive monolithic regulators often drive complex, discrete, active electronics in some designs. On the other hand, very complex discrete regulators often are used to drive low-cost op-amps. This approach is common in high-end Japanese designs. One interesting feature in the high-end Japanese designs is the use of a complete push-pull output stage in the regulator. It is unclear why a positive regulator should ever be required to sink current. One assumes that it has a role in reducing transient noise signals on the supply line.

Sometimes high-end discrete regulator stages use no global feedback. This approach, perhaps an attempt to mimic the low-feedback design of the active stages, results in a less capable regulator with much higher output impedance. Some high-end designs will be dual-mono right to the power cord. Other designers will use supply rails for both channels derived from a single voltage regulator.

Perhaps the greatest deviation in power-supply design occurs in power amplifiers. Some amplifiers will have complete regulation of all active elements, including the output stage. Holding the output-stage voltage rails constant is counterproductive if a purely engineering analysis is applied. A much more logical approach would be to dynamically vary the power supply voltage to the output stage so that the V_{BC} (or V_{DS}) of the device could be held constant. This would linearize the output stage and allow for the use of faster devices with lower V_{BCO} (or $V_{DS(max)}$). Despite the technical quicksand that power amplifiers with regulated output stages stand on, the open-loop sonic descriptions have often been very favorable.

Most power amplifiers do not have regulated output-stage supply rails, but some have regulated rails for the voltage-gain stages. This approach improves distortion performance, reduces crosstalk if the power-amp channels share the same power supply, and increases immunity to power-line noise. The downside of regulating the voltage-gain stages is a significant increase in complexity because the regulated voltage must be higher than the unregulated voltage applied to the output stage if the available output-voltage swing is not to be limited by the regulated supply. This requires additional transformer windings for the regulated power supply. Again, so-called "good-sounding" power amplifiers use no consistent power-supply design.

8. Exotic materials. Parts like 10 µF polypropylene capacitors and Teflon boards are just a few of the strange things that may be found in a high-end audio design. Other weird stuff may include silver wire and very expensive bulk metal resistors. These parts are justified on the claim that they sound better, although some measured differences have been reported [Curl and Jung 1985]. Many designers impose the requirement on themselves that the circuit must be flat to subsonic signals. Often these designers will not place electrolytic capacitors in the signal path [Jung and Marsh 1980]. As a result, parts cost for capacitors can become very high. Often a mix of electrolytic and film capacitors is used to reduce cost. The electrolytics are typically in those places in the circuit that would be most sensitive to capacitor nonlinearities, such as feedback loops. It appears illogical to assume that film capacitors in the less sensitive locations will improve the sound of the circuit if the electrolytics do indeed affect sound quality. One way around this is to eliminate coupling capacitors altogether. The dc offset is reduced with trim pots or active circuitry in the feedback path [B. Clark 1982]. Sometimes circuits are claimed to be direct-coupled because capacitors have been removed in the noncritical direct paths, but electrolytics are still found in the sensitive feedback loops. Such designs have received rave reviews from audio journalists whose ears are heavily influenced by simplistic technical claims.

Is there a conclusion to be arrived at here?

The examination of the topologies of audio equipment said to "sound good" has shown little commonality in the designs. Some designs use no feedback, others a small amount, and yet others a large amount at low frequencies. Some designers include the output stage in the feedback loop; others do not [Dalzell 1995]. Some designers use no capacitors in the signal path; others use only expensive film caps, while still others use less expensive electrolytics. Some designers use complex power-supply regulators, while others use no regulation at all in power amplifiers. Some designers will work mostly with FETs; others use only bipolar devices. Some designers use fully complementary circuits, while others use only single-ended circuits that are the latest vogue. Some designers use ICs in the signal path or for voltage regulation; others only use discrete designs. Some designers may even mix design styles within a given unit.

The random nature of the designs strongly suggests that no "X factor" parameter is being optimized. Instead, we can assume that the designer, using open-loop listening tests, has convinced himself that the changes he is making to the circuitry are affecting the sound. In this process—design, listen open-loop, design—the circuit designer has no checks and balances to guide him in his work. Open-loop listening is to a very great extent subject to the biases of the listener, and a designer wanting to prove that his new idea is better-sounding is clearly biased. Controlled blind listening tests would show if a sonic change were truly happening when a circuit change was implemented, but designers are unhappy when a new circuit idea is shown by such tests to be of no consequence. They thus try to dismiss the controlled test results, instead of facing the reality that electronics exhibiting proper measurements are sonically transparent.

This can often work in reverse. A designer might not use in his circuit an expensive component that would result in a measurable change in the device's performance because he has convinced himself through open-loop listening tests that the better component produces no sonic change. That is probably the reason why a lot of very expensive high-end equipment uses inexpensive D/A converters or digital interpolation filters. Considerably less expensive mainstream components, often said to sound less good in open-loop listening tests, use much better-performing parts.

The results of the study of the circuits discussed above confirm the results of controlled double-blind tests, which have shown that no sonic differences exist in audio electronics that measure well. When double-blind listening test are performed, random answers occur to the question "Which component sounds better?". When circuits that are claimed to "sound better" are analyzed, random design techniques are noted. Both analysis techniques, approaching the subject from opposite ends, converge to the same conclusion: audio electronics that measure properly will sound acoustically transparent. No "X factor" exists. Designers are wasting their time developing audio equipment using the "design, listen, design" approach because they are not using controlled techniques in the "listen" part of the process. If controlled techniques were used, the designers would discover that audio design is no different form other electronic design. It is done with a set of specifications, with paper and pencil, with computer analysis programs, and with laboratory measurements.

It should also be noted that audio—high-end audio in particular—appears to be the only technological discipline suffering from the peculiar attitudinal syndrome analyzed above. You will not find automotive engineers, for example, claiming that one brand of spark plug (or ignition wiring or distributor) with exactly the same specifications and measured performance as another "feels better" when driving, and certainly not that it makes the car go faster!

We are faced with so many real problems in audio design. It is time for the designers of audio electronics, when they are not accomplishing anything, to recognize it and move on to the solution of those real problems.

Goals for the audio designer.

In the final analysis, the designer of audio equipment should attempt to design the most transparent circuit consistent with cost goals and reliability requirements. Careful analysis of the literature and competitive designs will help him in this goal, as will controlled listening tests.

We believe that the design process can be relatively straightforward. Our recommendation is to design an audio product in such a way that its measured performance exceeds the target specifications. In this case the target specifications come from psychoacoustic studies of human hearing. Value added can be achieved by bringing out a significantly overdesigned product that well exceeds specifications, offering enhanced build quality and improved reliability. It is not the designer's job to worry about the delirious condition of the consumer audio industry. That should fall to the marketing department, which must come up with a plan to sell a well-designed product in a difficult market. In consumer audio the marketing department may have to work harder than the designer. That might be something of a first.

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(continued on page 41)

An Assortment of Conceptually Original Loudspeaker Systems

By Peter Aczel
Editor and Publisher

There are me-too speakers and there are me-different speakers. Both kinds can be good, but the best performers are usually a little more creative in concept than the plain-vanilla designs. As follows.

I have touched upon this before, but it needs to be repeated frequently—that no loudspeaker system, no matter how brilliantly designed, can be better than the theoretical model of the ideal electroacoustic transducer it tries to emulate, and that there is more than one such theoretical model. The arguments about point source vs. line source, monopole vs. dipole vs. bipole, wide dispersion vs. controlled narrow dispersion, flat frequency response vs. flat power response, minimizing reflections vs. deliberately introducing reflections, etc., etc., are endless and still unresolved. Thus, the world's most perfectly executed point-source speaker may not be as good as a slightly flawed but basically well-designed line-source speaker—if indeed the line-source concept is inherently superior, which it may or may not be. And that's just one example.

Then there are other considerations, such as the speaker's dynamic capability and distortion characteristics, which may or may not be in conflict with the choice of the ideal wave-launch model. Also, as I have pointed out several times in the past, testing the speaker's performance is not nearly as cut-and-dried as measuring an amplifier or a CD player. The latter have one input and one output per channel. A speaker has one input but an infinity of outputs. Which of these outputs do you compare with the input to test the speaker's accuracy? And even if the answer is, "As many of them as possible," how many are enough? Needless to say, the best minds in audio have sensible, practical answers, choices, and recommendations with respect to these various dilemmas, but there is certainly no such thing as the one design that satisfies all criteria of perfection. The best one can say about a loudspeaker design is that it is based on a highly credible concept and that the engineering implementation of the concept has total integrity. Are there such speakers? Not many, and they aren't cheap.

It should also be pointed out, to head off any kind of apples-versus-oranges controversy, that in stereo playback the apparent size of the soundstage at a given listening distance depends on the wave-launch geometry. A tall planar or line-source speaker puts the listener in the farfield even when he sits fairly close. A point-source speaker needs some distance to allow the spherical wave front to expand to quasi-planar farfield proportions. Thus there will be considerable differences in soundstage perception at typical listening locations in typical rooms. Such differences should in nearly all cases be attributed to the wave-launch geometry rather than the crossover configuration or the presence/absence of "time alignment" or any other design feature.

Hsu Research HRSW12V
(Part II)

Hsu Research, 14946 Shoemaker Avenue, Unit L, Santa Fe Springs, CA 90670. Voice/Fax: (310) 404-3848. HRSW12V powered subwoofer, $850.00 each (factory-direct, including shipping/handling). Tested sample on loan from manufacturer.

This was an unfinished review in Issue No. 23. Meanwhile (yes, I know, it's a long meanwhile between issues) the improved Model HRSW12Va is supposed to be almost ready to ship, and we have been promised an early sample. The 12V is still the current model of this outstanding powered subwoofer as I am writing this but on its way out. The coming improvements are said to include an even more advanced 12-inch driver and two large ducted ports instead of one, with flares terminating the ducts (to eliminate port-noise complaint—sorry Mr. Roth).

My measurements of the 12V yielded excellent results, as expected. The small-signal frequency response of the system, with the dedicated outboard amplifier in the loop, is dead flat (±1 dB or better) down to 20 Hz and -3 dB at about 17 Hz. The contribution of the equalized amplifier to this response is an 8.5 dB boost at 20 Hz. The amplifier is designed to deliver 150 watts into the 3.2Ω driver and is rather low in distortion: -83 dB (0.007%) just before the onset of clipping at 20 Hz, the maximum-gain frequency.

The acoustical distortion of the 12V is also quite low. I measured THD + N in the nearfield, placing the microphone in the floor-loaded space right under the speaker, at 1-meter floor-plane SPLs of 90, 96, and 100 dB. The lowest distortion was around 60 Hz (0.2% to 0.3% at all levels) and the highest around 24 Hz (1.4%, 2.3%, and 3.7% at the three rising levels). At 100 dB the 20 Hz distortion was 2.5%. I am fully aware that a slightly different methodology would have yielded slightly different results, but these figures show the general trend.

It is a reasonable expectation that the 12Va will be even better, but the 12V is certainly an outstandingly fine sub woofer, fully competitive with costlier high-end units. I have the greatest respect for Dr. Poh Ser Hsu, both as a resourceful technologist and as an audio manufacturer of integrity. He wouldn't know how to be a tweako/mystico high-end sleazoid if you held a gun to his head. Trust him to give you the best possible bass in the simplest and most cost-effective package. Just make sure you have the space—even his smaller subwoofers are pretty big.

*[**Flash!** Long after the above was written and shortly before this unconscionably delayed issue was ready to go to press, the 12Va arrived. It is indeed slightly better in every respect, including the amplifier, but still basically the same design, exactly as that lowercase* a *indicates. If you already have a 12V, don't consider yourself deprived, especially since the price has also gone up with the performance. Followup in the next issue.]*

JosephAudio RM7si

JosephAudio, 2 Pineridge Road, White Plains, NY 10603. Voice: (800) 474-4434. Fax: (212) 724-2509. Model RM7si 2-way minimonitor loudspeaker system, $1299.00 the pair (rosewood finish $200.00 extra). Tested samples on loan from manufacturer.

The "Infinite Slope" technology is back. Remember the JSE speakers from the mid-1980s? They were based on the same crossover design, conceived and patented by Richard Modafferi, our recently recruited RF consultant. (Yes, he knows filter theory like the back of his hand. Yes, this is a serious engineering concept, not a marketing gimmick.) For reasons that had nothing to do with the inherent merit of the design, JSE went out of business. Now JosephAudio (headed by Jeff Joseph) has taken on the cause and is determined to put the Infinite Slope crossover back on the map again. With Rich Modafferi's help, the company has developed new and significantly improved models, of which this one represents the bottom of the line.

The Joseph RM7si consists of a bookshelf-size (15 by 9 by 11 inches deep) vented box, a 1-inch silk-dome tweeter, a 6½-inch woofer with fiberglass cone and "phase plug," and the Infinite Slope network. See David Rich's sidebar for an explanation of how the crossover works. Here I shall restrict myself to the performance of the speaker.

My alleged curmudgeonly inclinations notwithstanding, I am always happy to come across an audio device that performs in every way as represented by its makers, and this is one of them. The RM7si may be a bit pricey for a small speaker with OEM drivers, but the sophisticated filter technology it incorporates probably justifies that, and the sound it produces is truly excellent. I cannot think of any comparably small speakers in my recent, and even not so recent, experience that pleased me as much.

The quasi-anechoic (MLS) frequency response of the speaker, taken at 1 meter, shows basically flat characteristics (±1.5 dB) for both woofer and tweeter, except that the woofer appears to be set 2 or 3 dB higher than the tweeter. I couldn't tell for sure whether this was deliberate "voicing" or some sort of step in the response of the 6½-inch driver itself. The Vifa silk-dome tweeter (also used in a number of ultrahigh-priced speakers) remains dead flat right out to 20 kHz. The crossover frequency is approximately 2 kHz. The filter skirts are very steep indeed but not "infinite"—42 dB per octave was the steepest I could read with an admittedly crude electroacoustic nearfield measurement. The vented box is tuned to 41 Hz, and the maximum output from the ducted port is at 52 Hz. The summed response of woofer and port is dead flat (±0.25 dB) down to 55 Hz and -3 dB at 43 Hz. There is still useful output (-12 dB) at 33 Hz This is very respectable bass response for a 6½-inch driver in a box of this size. The box itself is very solidly built and appears to be quite dead when knuckle-rapped.

The impedance curve of the RM7si is unusually flat in magnitude above the box-influenced frequencies, varying only from 7Ω to 12Ω in the entire audio range; the phase variations are also small, with ±20° the biggest swings above the tuned-box range. I bet Rich Modafferi did that on purpose, to allow you to use just about any half decent amplifier to drive the speaker. (In fact, David Rich connected our test samples to his bargain-basement Pioneer SX-203 receiver—which he reviews in this issue—and obtained the same audible results as he did with his big-bucks amplification system.)

I took distortion readings only on the woofer, since I know from previous experience that the Vifa silk-dome tweeter has entirely negligible distortion in the range used here. Over its midrange band, namely 200 Hz and above, the woofer stays in the 0.2% to 0.35% THD range at 1-meter SPL's of 90 to 95 dB (I generally don't mea-

What Exactly Is an "Infinite Slope" Crossover?

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

It is time for our lesson in network theory. Today's talk is on poles and zeros. I am not going to tax your understanding with complex transfer functions in the s-plane and stuff like that. I'll just treat poles and zeros as system elements that cause declines and rises in frequency response. As you recall from last time, a simple pole will cause a rolloff at 6 dB per octave, and the signal will be down 3 dB at the pole frequency. You will also recall that a complex pole pair has a pole frequency and a Q. The Q tells us the damping in the system. No damping and the system oscillates. Now the frequency response of a complex pole pair rolls off at 12 dB per octave, just like two simple poles, but its action around the pole frequency is very dependent on Q. For a Q of 0.5 the response is just like two simple poles of the same frequency. For a Q of 0.7 the response is down 3 dB at the pole frequency, and no peaking occurs. Beyond a Q of 0.7 we get peaking. For high Q's we can get very large peaking.

Now, in today's lesson we learn about zeros. Zeros are the opposite of poles. A zero that is not located at 0 or °° is called a finite zero. A simple zero rises at 6 dB per octave and is up 3 dB at the zero frequency. Complex zeros rise at 12 dB per octave, with the response around the zero frequency again determined by the Q. Just as in the case of a complex pole, Q = 0.5 when the response is like two simple zeros of the same frequency. For a high Q we get a dip at the zero frequency, the opposite of the peak we had with the poles. If the Q is infinite, the dip goes to a null point where no signal gets through. Infinite-Q zeros are still stable and usable, unlike poles. (Remember that, because it will be on the final and lots of people are going to get that question wrong.)

A simple highpass circuit has all its zeros at 0. That's why the response starts rising from dc. When the frequency exceeds the pole frequency, the response flattens out as the falling response of the poles cancels the rising response of the zeros. A simple bandpass circuit has half its zeros at 0. The response starts rising from dc. Since twice as many poles will be crossed as the number of zeros we started with, the response starts falling at the same rate at which it was rising once the incoming frequency exceeds the pole frequencies.

As you recall from a pervious class, we can place poles to give us sharp filter responses, using a variety of pole placement methods. Bessel gives good time response, Butterworth is the flattest response you can have, and Chebyshev has a sharp transition band but at the cost of passband ripple. But what happens if we add zeros to the mix? In a lowpass filter, if we add a pair of imaginary (infinite-Q) zeros above but near the transition band, then the transition band will be very steep as the signal falls into the infinite dip caused by the zeros. Since the zeros will increase signal level past their zero frequency, the price to be payed for using the zeros is that the stopband will not roll off as steeply. Indeed, if the number of finite zeros equals the number of poles, the response will flatten out after the transition band. But if we are down 40 dB or 60 dB, does it really matter that we will go no lower?

Using finite zeros in filters is not a new idea. It was developed many years ago at Bell Labs. Every telephone call you make goes through filters that have finite zeros. So why do we not use zeros in crossover designs? Well, you recall from that long homework assignment I gave last week that the synthesis of passive filters is much more difficult than active-filter synthesis. Adding finite zeros makes the passive synthesis problem even harder. In another long homework I gave you, I required you to match a highpass and lowpass filter so they summed to flat frequency response. Remember all the trouble you had figuring how to stagger the highpass and lowpass sections? Think how hard that would be if we added finite zeros to the mix! Also, you will all recall from my lecture on sensitivity that component variations will change the filter's response, causing all sorts of ripples, and the addition of finite zeros will make this worse. *[Hey, Dave, what if the guy reading this is not an undergraduate E.E. student type but a salesman in a cream-colored polyester suit trying close a sale on a JosephAudio speaker?—Ed.]* And since the transition band is steeper when we add in finite zeros, the time-domain response is going to be worse also.

Well, if you have a very good understanding of network theory, have a good knowledge of computer optimization and are very clever, you can use finite zeros in a passive crossover network. Of course, the aforesaid job requirements are not going to be met by your typical loudspeaker-company designer. It took someone called Richard Modafferi to figure out how to make a practical crossover network with finite zeros. The method of synthesis is so clever (it uses a transformer, among other things) and elegant that it is patented, and Rich derives an income from the royalties. Now, the slope of the Modafferi crossover is not "infinite" but it goes down real fast. It comes back up real fast too, but the marketing department never reports the response that far out, so most people assume that the response drops to infinity and stays there forever. Never mind the marketing mentality; despite the hype and the fudging of the curves, this is a major advance in crossover design. I will discuss how to deal with marketing people in your design class. Time's up; class dismissed; let's go listen to some loudspeakers.

sure at unbearably high sound pressure levels). Below 200 Hz the THD climbs rapidly but is still in the 0.25% to 1.4% range at 90 to 95 dB, right down to 57 Hz, at which point it shoots upward and passes the 10% mark at 40 Hz. Again, this is quite respectable (and then some) for a bookshelf-size unit. If you were to cross the RM7si over to a subwoofer, I would suggest a crossover frequency of 80 to 90 Hz, where the speaker is still very flat and the THD is low.

The audible outcome of all these good design characteristics is very favorable; the speaker has a basically neutral tonality, with very smooth highs, a thoroughly transparent midrange, and a surprisingly solid, satisfying bottom end. The soundstage is open and plausible; the imaging is all it should be. The total noninterference of the two "brick-walled" drivers is clearly an advantage, with no negative aspects that I can discern. (All you first-order crossover cultists take note—maybe you've got it all wrong, huh?) The totally low-end-protected tweeter can be played louder, and the absence of interference patterns in the crossover region appears to remove a layer of veiling that I often hear in speakers with lower-order crossovers. So—what we have here is indisputable high-end loudspeaker sound in a compact package, at a price still well below the loony category. Only the deepest bass is missing, but you know that coming in, just by looking at the size of the box. The Joseph Audio RM7si is therefore highly recommended.

Newform Research R8-1-30

Newform Research Inc., P.O. Box 475, Midland, Ont., Canada L4R 4L3. Voice: (705) 835-9000. Fax: (705) 835-0081. Model R8-1 floorstanding 2-way loudspeaker system with R30 ribbon, $2095.00 the pair (list price) or $1236 the pair (delivered direct from factory). Tested samples on loan from manufacturer.

The concept implemented by this ribbon-based loudspeaker system is the monopole line source. Designer John Meyer has developed 8-inch and 15-inch ribbon modules consisting of a very narrow (0.75-inch) film diaphragm, tightly suspended in a magnet structure and blocked in the rear. The modules are mounted end to end to form upper-midrange/treble drivers of different dimensions. The model reviewed here incorporates two 15-inch modules forming a 30-inch ribbon, mounted as a freestanding pole on top of an enclosure housing an 8-inch woofer and a ducted port made of granite. The crossover is at approximately 1 kHz. The highpass filter appears to have a slope of 18 dB per octave, the lowpass 12 dB per octave (these being the combined results of the electrical and acoustical components of the crossover).

John Meyer claims "the deepest and most focused soundstage in the business," as well as "openness, transparency, smooth frequency response and wide bandwidth," for the Newform ribbons. I can confirm that the soundstage is as precisely defined and the overall sonic presentation as open and transparent as any audiophile could possibly wish for. The tall, narrow wave-launch source works as it is supposed to. Furthermore, the bass reproduction is remarkably good for a vented box using only a single 8-inch driver. And, let's not forget, the price is right. Then why am I not happy?

On every sizable dynamic peak involving the treble range, the speaker turns harsh, edgy, nasty—and my ears go ouch. It is not a subtle, debatable effect but an obvious characteristic that will bother some listeners more than others. It will probably be tamed to some extent by a very dead room with minimal reflections at the higher frequencies; my listening room is not as dead as all that, although quite a bit deader than typical contemporary living rooms and family rooms. For my own use, this fault disqualifies the speaker, even though its many virtues made me initially root for it. It is a fault, however, that not every audiophile appears to be sensitive to.

My quasi-anechoic (MLS) measurements of frequency response were taken at a distance of 2 meters, since the speaker is too tall and spread-out for the standard 1-meter setup. The overall response is indeed smooth, as claimed, but not flat. There is a preponderance of energy in the two octaves between 1 kHz and 4 kHz, almost regardless of how high or low the calibrated microphone is aimed. This could very well account for the aggressive edge on dynamic peaks. From 4 kHz to 20 kHz, the response is reasonably flat, with a slight downward trend that flattens out in the top octave (10 kHz to 20 kHz). The ripples in the response shift around considerably as the height of the microphone is varied. (A point source is certainly easier to measure.) On the other hand, at 30° *off axis* horizontally, the response is barely different from one at the same height *on* axis, thanks to the narrow line source. The theory is confirmed.

Bass response was measured with the nearfield method. The summed output of woofer and vent is flat within better than ±1 dB down to approximately 40 Hz; the -3 dB point (f_3) is at 33 Hz. You can't do better than that with an 8-inch driver in a box of acceptable size and still have adequate efficiency left. The tuning of the vented box appears to be somewhat unconventional, as there is no clear null in the output of the driver (only a clear peak output from the vent at approximately f_3), and the bass rolloff slope is only 12 dB per octave. Today's bass alignment software offers more possibilities than I can sort out.

Distortion measurements (THD + N versus frequency) were taken right off the woofer cone and the ribbon, with 1-meter SPL readings normalized to 200 Hz and 2 kHz, respectively. The woofer at approximately 90 dB and 94 dB varied very little in distortion, which was never lower than 0.25% nor higher than 1% down to about 80 Hz. Below that the distortion rose rapidly, as it does in nearly all conventional designs. At 30 Hz, it reached the 4% to 5% area, which is normal. The ribbon also drew rather similar curves at 90 dB and 94 dB, but these were much more like a roller coaster. Above 8 kHz the curves shot skyward, reaching 3% at 10 kHz and still climbing. Minimum distortion at the levels tested was in the 3 kHz to 8 kHz band, where it remained between 0.25% and 0.8%. At approximately 2 kHz there was a big distortion peak, 2.3% at 90 dB and 4% at 94 dB. (A further confirmation of the "ouch" experience?) In the crossover region the distortion dropped to about 0.7%. All in all, these figures are not as good as can be obtained with the best conventional speakers.

I also detected some ringing at 3.5 kHz as I explored the ribbon with tone bursts. Nothing dramatic but quite noticeable and well within the "ouch" range. Of course, that kind of correlation doesn't prove cause and effect, but then I'm only a reviewer, not the R & D man for Newform.

As for impedance, the magnitude swings between 6 ohms and 30 ohms (the latter only at the crossover), the phase between -30° and +45°. That's not exactly like a resistor, but it's a load that doesn't require a high-current amplifier, only a reasonably stable one.

Can I recommend this loudspeaker system? Only if you value soundstaging, focus, and transparency over easeful dynamics, sweet tonality, and low fatigue. If that's where your head is at, the Newform Research R8-1-30 is very good value. To me, as I said, its shortcomings outweigh its virtues.

Signet SL256

Signet, 25 Esna Park Drive, Markham, Ont., Canada L3R 1C9. Voice: (905) 474-9129. Fax: (905) 474-9812. SL256 compact 2-way loudspeaker system, $360.00 the pair. Tested samples on loan from manufacturer.

I am always looking for low-priced speakers of credible design and respectable performance, but they are hard to find. To the best of my recollection, I have never reviewed any speakers as low in price as the Signet SL256 from Canada, not even back in the relatively uninflated 1970s. I thought I'd give the SL256 a try because it appeared to duplicate all the basic design features of higher-priced compact speakers. Such a speaker is made possible today by the availability of good mass-produced OEM drivers.

The trouble is that I have the high-end speaker sound in my ear at all times and find it difficult to think in terms of per-dollar sonic quality. The SL256 sounds quite decent as far as tonality is concerned; it is never harsh or overbright or otherwise unpleasant; the music sounds like real music; but there is a hollow, congested, boxy quality, not very obtrusive but present on all program material, that says "cheap speaker." Is that par for the course at $180 per side? I really don't know.

The speaker consists a 6½-inch polypropylene-cone driver and a -inch soft-dome ferrofluid-cooled tweeter in a small vented box (ducted port in the rear). The crossover is at 3 kHz. Thus all of the fundamental tones of music are assigned to that one 6V2-inch diver, and only the overtones are handled by the tweeter—not exactly a recipe for perfectly balanced, uncolored reproduction. No cone driver can cover six octaves with equal quality. The frequency response of the SL256 is quite rough; the 1-meter quasi-anechoic (MLS) measurement cannot be interpreted, even with the greatest goodwill, as better than ±4 dB up to 20 kHz. There is a wide, ragged dip from 1.3 kHz to 6.4 kHz, about -5 dB at its deepest, which may be "voicing" to prevent edginess. The vented box is tuned to about 54 Hz, and the bass response starts rolling off in earnest at around 45 Hz. I measured distortion only at a 1-meter SPL of 90 dB, where it remains below 0.5% down to 140 Hz, at some frequencies even quite a bit lower than that, but rises rapidly in the bass. The impedance curve is a roller coaster: 4.6Ω to 55Ω (at the crossover) in magnitude and -55° to +55° in phase.

I can't work up too much enthusiasm for this design; on the other hand, I must admit that the speaker is very attractively priced considering the fairly decent driver complement, the solidly built cabinet (surprisingly dead in response to knuckle rapping), and little quality touches such as felt padding around the tweeter. I suppose this is what you can expect to get for your money from a good speaker company.

Sunfire "True Subwoofer"

Sunfire Corporation, 5210 Bickford Avenue, Snohomish, WA 98290 or P.O. Box 1589, Snohomish, WA 98291-1589. Voice: (206) 335-4748. Fax: (206) 335-4746. "True Subwoofer" 2700-watt powered subwoofer, $1250.00 each. Tested samples on loan from manufacturer.

When Bob Carver enters the arena in a product category entirely new to him, you can be sure of something highly original and creative. He would rather die than come out with a me-too audio component. It should be no surprise, therefore, that the Sunfire "True Subwoofer" is a tour de force. I don't think anyone else in the audio community could have designed it. Certainly none of the tweako lightweights who gather at every high-end event and act as if their mutual admiration society were conversant with a higher truth than the Bob Carvers of this world could possibly fathom.

I don't want to create the impression that I believe this is the best subwoofer of all time. Far from it. It is merely the smallest, by far, as well as the most ingenious, by far, of all the subwoofers in existence that deserve audiophile consideration. Now you can have high-SPL

Inside the Sunfire "True Subwoofer"

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

In Bob Carver's original concept for this subwoofer, he was going to supply the amplifier power necessary to make an 0.8-cubic-foot speaker—outside dimensions, not volume!—produce 20 Hz at high SPL. The problem was that the amount of power required could not be dissipated in the voice coil. In a typical driver, 99% or more of the power is dissipated as heat; only 1 % is transformed to acoustic energy. Greatly increased power means too much heat to be dissipated in a limited space; it cannot be dealt with. The efficiency of the driver had to be increased. But how?

The 10-inch driver was as big as one could get in the box. Bob was already using a passive radiator to improve the efficiency, and increasing the box size was not negotiable. What he discovered was that if he used a really huge magnet he could increase the woofer's efficiency, but there was a price. The large field increased the back EMF (the voltage that results as the voice coil cuts the magnetic flux in the gap of the magnet). All loudspeakers have back EMF, but the giant magnet made for a very large value. The back EMF voltage is 90° out of phase with the current flowing through the voice coil. That means no power is dissipated at the loudspeaker. At the power amp it is another mater. The voltage difference between the power-supply rail and the speaker-terminal voltage, multiplied by the current being sent to the loudspeaker—that is the power that must be dissipated by the amplifier's heat sink. For this reason, an amplifier gets hotter when driving reactive loads than when driving a pure resistance. With a purely resistive load the big current flow occurs when the output voltage is high and the drop across the power-amplifier output device is low. With reactive loads you can get high current and high voltage across the output devices at the same time. Because of this problem, back EMF is limited in typical production loudspeakers in a tradeoff against efficiency. Just like your power company, power amp designers do not like driving loads with bad power factors.

But Bob Carver had the solution to the power-supply dissipation problem: use a tracking power supply and keep the voltage across the output devices low. The same technology allows him to build a power amplifier with very large voltage swings. Even though the amplifier can drive the load to very large voltages, the output devices never see that voltage because the power supply tracks the output. So Bob had what he needed—a woofer that was efficient but one that required a very high working voltage, plus a power amplifier to create the voltage. Most of the voltage was dropped across the lossless back EMF, but it had to be sourced by the power amp. And Bob had just the power amp to do the job.

The electronics of this unit are very complex. The first thing to understand is that the amplifier is line-powered, yielding rail voltages of 165 V. The input signal circuitry must be isolated from the line. A digital optocoupler does this. Driving a digital optocoupler with an analog signal results in distortion. The fixed single-pole highpass filter at the Hi-Pass Out jack of the subwoofer is unbuffered. The input impedance of the satellite power amp will move the pole. This highpass filter is all but useless. The lowpass filter is a sixth-order design. The filter passband is varied by changing one component in one section of the filter. As a result, the effective width of the transition band changes with the cutoff frequency. It would take a six-gang potentiometer to move all the poles at the same time and keep the transfer curve constant. A phase-shift circuit with trim pot is also in the signal path, and so is a second-order subsonic filter. A sharper highpass filter than the latter, with a 29 Hz cutoff, can be evoked by activating the THX switch. This would be nice for smaller rooms, but the THX mode also boosts the bass before the cut occurs.

A master protection circuit consisting of eight op-amps and fifty passive components monitors overshoot, clipping, excess low frequencies (what the subsonic filter failed to attenuate sufficiently), thermal runaway, and excessive excursions. The power amplifier itself is essentially similar to the Sunfire power amp. The supply rails are higher but current-sinking capability is lower (we know the load we are driving). Three parallel MOSFETs on each downconverter drive a pair of output devices per side. The amplifier itself is as simple as it gets—Pioneer SX-203 simple. A single differential pair has a resistive load and a resistive current source. The second gain stage is loaded by a bootstrapped resistor. This is a circuit that brings back memories of the '60s amplifiers, including the original Phase Linear amplifier. This circuit trick allows the second-stage output to move above the supply rail, so the output stage can be driven with more cur-

bass all the way down to below 20 Hz, the kind you wish you had the space for, out of an 11-inch cube, complete with electronics. That's 11 inches, i.e., 28 centimeters, in all three dimensions, plus a little bulge on two sides where the active and passive radiators are. Amplifier included; all controls and terminals on a third side. Now then, tweaks, could Dave Wilson have done that? Could Arnie Nudell? Yeah, sure.

How did Bob do it? That's a long story. He tells it in an elaborate "white paper," obtainable for the asking from Sunfire Corporation The paper, which I have so far seen only in its 13-page draft version, is full of math rent on positive excursions. Modern amplifiers use active pullups. The output of the second gain stage drives one set of predrivers, which then drives the output stage. No go-fast circuit is present in this implementation, since the input is bandlimited to 100 Hz by the sixth-order low-pass filter. (That's why Bob gives us such steep slopes.) Low distortion is not the long suit of this amplifier design.

The Sunfire amplifier is perfect, however, for this application. No RFI (radio frequency interference) problems from the switching regulators in the downconverters occur because the speaker signal never gets out of the box. One should note that a class D amplifier would also work well in this application (the tracking downconverters connected directly to the load) because of the bandlimited nature of the subwoofer signal. Construction quality of the system is nice. The boards are double-sided and filled with reliable if not exceptional components. For example, the op-amp in the signal path is the Motorola quad MC34004. The signal goes through 11 stages of the MC34004 before it gets to the amp. The build quality of the electronics is closer to high-end US stuff than Far East mass-market items.

The sixth-order filter in this subwoofer actually gives you a chance to blend its output with that of a ported woofer in the main speaker. A hole or peak will occur in the transition band, but it will be narrow and, since room response is far from flat below 100 Hz, probable not noticeable. This works especially well if the main speaker has some bass extension. Minimonitors are more problematic because the crossover must occur at a higher frequency. Sealed systems present a bit of a problem. They need to be highpass filtered. As stated above, the Sunfire fixed filter is pretty useless for this. You can do a roll-your-own 6 dB per octave filter between the preamp and power amp if you know your power amp's input impedance, but you may need a higher-order crossover. If this sounds like you are being asked to become a speaker designer, you are correct.

Let's get back to the case of a full-range ported speaker. I would recommend the following procedure. You will need a test CD with low-frequency warble tones. I used the *Stereophile* disc. (Why do I hear screams from the Editor?) You will also need a sound-level meter. The cheap Radio Shack unit is just fine.

Step 1. Disconnect the subwoofer and run the main speaker with a tone in its passband (80-100 Hz). Measure the level.

Step 2. Disconnect the main speaker and reconnect the subwoofer. Set the subwoofer to its highest crossover frequency. Set the level control of the subwoofer to give the same sound pressure level with same tone you used in Step 1.

Step 3. With both the subwoofer and the main speaker connected, measure the level of the tones at the available frequencies. Because the crossover is set too high, you will have a peaked response. Adjust the crossover control to get the smoothest response.

Step 4. Use the phase control to make the response even smother. It has its biggest effect at the crossover frequency. You can iterate between the crossover and phase controls. Keep your hands off the level control ! It was set correctly in Step 2.

Step 5. Listen to the subwoofer. Resist all temptations to turn up the level control. Play something with really deep bass to confirm that your subwoofer is working. On normal recordings you will have very little coming out of the subwoofer—an occasional double-bass growl or a deep percussion instrument. These little outputs do count because the overall foundation of the orchestra is brought up in level. It is a small but perceptible difference. Is it a $1250 difference? That is something for you to decide. *[Dave, are you aware that people buy subwoofers to hear the low frequencies in* Jurassic Park, *not in Bruckner?—Ed.]*

The Sunfire True Subwoofer is small enough for anyone to add a subwoofer to any room. That is a major advance in the state of the art of such designs. Many (including me) who do not have listening rooms large enough to allow a normal subwoofer with 20 Hz response to be brought into the room—whether the room will support such a frequency is a separate issue—will find the Sunfire True Subwoofer makes this possible for the first time. I obtained good results matching it to a pair of Sound Lab Quantums (electrostatic panels with dynamic woofers). Less good results were had with the Monitor Audio Studio 6 minimonitors because of the higher crossover frequency. Minimonitors really need a dedicated crossover design if the bass quality is to be unaffected when matched to a subwoofer. For some, the addition of two octaves of bass will be worth the coloration of the midbass, but having the cellos muddied up to hear the basses better is not a trade I would make.

One bad feature for apartment dwellers is that the asymmetry of the two drivers causes a lot of floor shake. A good feature is that the amplifier will shut down when signal level drops below a certain low value. Because of the effect of the equal-loudness contours you are not going to hear the subwoofer at low levels, but people in adjacent rooms may well hear it, so it is nice that it turns itself off.

The True Subwoofer satisfies one of my audio fantasies: 20 Hz in the small rooms that real people live in. I am a very happy camper!

(nothing frightening, just a few derivatives and trig functions) plus a bit of Carveresque hyperbole, but it explains very clearly how and why a complete rethinking of conventional engineering precepts regarding the interdependence of bass extension, enclosure size, and efficiency leads to the conclusion that an "impossible" design like the Sunfire subwoofer is indeed possible and perfectly logical. David Rich, who is his own running white paper on most engineering subjects, tells the story from his perspective in the sidebar on pages 30-31. Here I shall deal only with the barest fundamentals. Rest assured, in any event, that the laws of physics have not been broken, only very artfully reshuffled.

Bob's basic insight was that you can keep making the enclosure smaller and smaller, thereby lowering the efficiency, and the woofer motor structure bigger and bigger, thereby counteracting the loss of efficiency but also creating humongous back EMF, until there is no amplifier that can drive such a load—except his! The Carver/Sunfire amplifier circuit with its tracking down-converter (see Issues No. 22 and No. 23) is capable of producing the very high-voltage output needed to overcome that large back EMF, yet it takes up so little room that it can fit into that 11-inch cube. Without the amplifier, which looks more like a computer motherboard, the subwoofer would not be possible. Luckily, the system is vastly more efficient than conventional engineering thinking would likely have predicted—that was part of Bob's contrarian brainstorm—so that the slimmed-down, heat-sinkless, high-output amplifier is more than adequate for the job. That's the story—a sweeping over-simplification, but did I promise you anything else? Let David explain the details.

The 10-inch (o.d.) driver is unlike any other; I'm almost certain that the OEM supplier's marching orders from Bob were something like "don't think, just do it my way." The magnet weighs almost as much as a men's shot put; the surround looks like a garden hose slit lengthwise and bent into a circle; the diaphragm is so thick and heavy you can sit on it; the excursion capability is huge. The passive radiator (this is a vented sytem) looks almost identical to the driver, except that it has an even more massive diaphragm and can actually withstand a solid blow with the fist (Bob's favorite demonstration). I have deliberately left out the numerical driver specs because (a) I am not in a position to verify them accurately and (b) they are meaningful only if you are familiar with the less impressive numbers of other bass drivers. It should be pointed out, however, that this is not the kind of overdesign intended to dazzle the wide-eyed audiophile; the truth is that with just a little less of said overdesign the system wouldn't work at all. What we have here is not a deluxe audio component but rather a bare-bones component just sufficient to do a high-performance job reliably. In that sense it is also in a unique category, in addition to the big-bass-in-a-small-box consideration.

On the input/output/control panel that takes up an entire side of the cube, there are three control knobs: volume, crossover frequency, and phase. You can juggle these until you get a good match to your main speakers, but remember that the crossover frequency is in effect the corner frequency of the steep lowpass filter—120 Hz down to 40 Hz—there being no crossover circuit as such. The highpass outputs jacks are a joke—passive circuit, series capacitor, 6 dB per octave attenuation, turnover frequency varying with load impedance. Definitely not for the control freak (maybe for very small satellite speakers with poor bass-power handling?). The variable lowpass filter has 36 dB per octave slopes in the stop-band, but in the transition band the Q gets lower and lower as you turn the control toward 40 Hz, at which point it doesn't look like a very steep filter anymore, until it gets past 100 Hz. Thus the overlap with the main speakers depends to a large extent on the setting of the control knob. Luckily a subwoofer crossover of, say, 70 or 80 Hz does not have to be super precise for listening satisfaction—I say luckily because a crossover that will mathematically reassure a student of filter theory can only be achieved, if at all, with a woofer fixed in position relative to the other drivers.

The frequency response of the Sunfire subwoofer is the sum of the outputs of the opposite-firing active and passive radiators, which are of the same size and look quite similar. With the unit on the floor, I found a very good acoustical summing junction just above the box, slightly closer to the passive radiator, at which point the microphone read an almost dead flat response: ±1 dB down to 18 Hz. With a more sophisticated measurement technique—vectorially adding the active and passive outputs, with compensation for radiator area if necessary—I think the response would have been amplifier flat. Below 18 Hz the response plummets like a stone; there is no "useful response down to" any lower frequency.

The distortion performance of the Sunfire subwoofer raises some questions. At frequencies down to almost 40 Hz, measured my usual way (away from the walls, on the floor, microphone right on the active diapragm), at my usual levels (90 to 100 dB SPL @ 1 meter), the distortion is quite reasonable: 3% or less second harmonic, 1% or less third harmonic. But—going down from the low 40s to 20 Hz (with the microphone on the passive diaphragm below 25 Hz), the distortion rises dramatically, even at these less than outrageous levels, reaching worst-case figures in excess of 10% second harmonic and 2% third harmonic. Now—hold your horses. The user's manual that comes with the subwoofer unequivocally recommends that the unit be placed in a corner of the room. That means it will radiate into one fourth the solid angle it sees when placed on the floor away from the walls, as it was for the above measurements. Needless to say, the efficiency goes way up with corner placement (which yields a quasi horn), the drive required for a given SPL goes down, and so does the distortion produced at that level. Indeed, with corner placement, we are back to (very roughly, give or take a bit) the 3% second harmonic and 1% third harmonic maximum readings, this time all the way down to 20 Hz, in the same range of levels. Once again respectable but not outstanding distortion performance.

Yes, I know your next question. Why should the Sunfire subwoofer have the advantage of being measured in the corner, when this journal routinely measures distortion at SPL readings away from the walls? The answer is simple: the Sunfire fits neatly into the corner, whereas the bigger subwoofers do not. The Sunfire has the inherent ability to take advantage of the quasi-horn effect of the corner; the bigger subwoofers much less so. Let each device be used to its best advantage. You're not satisfied with that answer? Then show me an 11-inch cube with lower distortion at the same SPLs.

As for the maximum SPL the Sunfire subwoofer can produce, it varies with frequency, room placement (i.e., loading), excitation time (since the limiters prevent prolonged excitation at SPL/frequency extremes), and the position of the Flat/Video-Contour toggle switch (higher SPL but less extended bass in the video position). I have verified that a single unit placed in the corner of my largest listening room (22 by 20 by 9 feet) is able to generate unbearable (to me) sound pressure in the 20 Hz to 40 Hz octave and rattle all the windows and shelves. I am not enough of a masochist to get involved in extensive measurements at triple-digit SPLs. I'll let Bob Carver and his competitors argue about the ultimate SPL capabilities of the Sunfire subwoofer; for the typical user it won't be an issue.

Are you now asking me how the Sunfire subwoofer sounds? That's almost as naive a question as how a well-designed amplifier sounds. A well-designed subwoofer sounds exactly the way it measures. The lowest frequencies are not at all mysterious. This subwoofer is flat down to 18 Hz, can produce very high SPLs, and has average distortion characteristics. That's precisely the way it sounds. If I blindfolded you and told you it was an 18-inch woofer in a 20-cubic-foot enclosure, you would probably believe me. If that's the kind of sound you want and you have no room for a big subwoofer, the Sunfire is the only game in town. If you have lots of room, you obviously have other options, but the unique high-tech chic of the Sunfire may still prove to be a fatal attraction. There is nothing else like it.

Waveform Mach 17

Waveform, RR #4, Brighton, Ont., Canada KOK 1H0. Voice: (800) 219-8808. Fax: (800) 219-8810. Waveform Mach 17 floor-standing 3-way loudspeaker system with electronic crossover, $5995.00 the pair (direct from Waveform, to be raised to $6995.00 in 1997). Tested samples on loan from manufacturer.

Before anything else, it should be pointed out that the price of this speaker would be closer to $14,000.00 the pair if sold through dealers instead of directly from the factory. Then I should add that no $14,000 speaker I'm aware of comes even close to it when all aspects of performance are considered. And then I should add... well... that no speaker I'm aware of, at any price (be it $14,000 or $140,000), equals it in transparency and lack of coloration—but now I'm getting ahead of myself.

As I pointed out above, every theoretical model of the "ideal" loudspeaker system has its nonnegotiable electroacoustic givens, and no speaker can be "better" than its theoretical model. In this case the theoretical model is the point source. The Waveform Mach 17 is in my reviewing experience the most highly perfected full-range electrodynamic speaker system based on the point-source concept. In many ways it is at the limit of what is possible with available drivers and the best textbook knowledge. That does not mean, however, that Martin-Logan or Magneplanar or Sound Lab or Carver "Amazing" fans will hail it as the cat's meow. It is, and sounds like, a highly accurate point source, not like a planar or line source. Imagine a truly superb minimonitor, much more dynamic than any other and extended on the bottom by a very fine bass system. That will give you a general idea of what to expect sonically.

The design is totally different from earlier Waveform models. About the only similarity is the truncated pyramid look. It is a big speaker, nearly four feet high, with a footprint close to two by two feet, but not so big as to fall automatically into the "over my dead body" category of spousal opposition. Furthermore, it disassembles into two manageable modules, each under the maximum UPS shipping weight. Piano movers will not be needed, but don't try to move it around too much before you've had your Wheaties.

The heart of the Mach 17 is the egg-shaped head module, which houses a 1-inch silk-dome Vifa tweeter and a rather unusual 7-inch (o.d.) Audax midrange driver with plastic cone, flat surround, and a rubbery-looking phase plug. The Audax was designed strictly as a midrange unit, not as a midrange/woofer for small systems. The tweeter is the current darling of speaker designers; the $19,000 Snell Acoustics Type A also uses it. Waveform gets it with a custom faceplate cut off at the bottom, to get it as close to the midrange as possible. What makes the Mach 17 unique, however, is the enclosure for these two drivers. It is made of special fiberboard laminations (16 layers!), lathe-turned into a beautiful egg shape, with about a quarter of the egg cut off flat to create a mounting surface. You've never seen anything like it. The idea is to minimize diffraction to the nth degree, the egg being definitely one up on the usual rounded box edges in that respect. What's more, such a laminated structure is totally dead acoustically. The egg is mounted on top of the woofer cabinet much the same way as a computer monitor, so you can tilt it up and down and swing it from side to side. The top of the woofer cabinet is padded with acoustic foam rubber to prevent reflections from the egg. The whole thing is so unusual that some owners will undoubtedly leave off the standard grille-cloth hood just to keep the egg exposed. It's so cooool...

The bass system consists of two Philips 12-inch woofers mounted in a vertical array and two 3-inch-diameter ducted ports. The cabinet has 0.75-inch walls and isn't exactly light, as I said, but when the head module is removed a retracting handle appears, providing a convenient handhold. John Ötvös, the owner and president of Waveform, is a master cabinetmaker and has thought of everything packagingwise. He made no attempt, however, to create a showpiece cabinet for the Mach 17, as he did for earlier models; the emphasis here is on solid craftsmanship and simple good looks, leaving more money for the engineering features. That was the department of Dr. Claude Fortier, physicist, acoustics professor, and circuit designer. He and John have been working together off and on for ten years.

One engineering decision was to have no passive crossovers anywhere in the system. They are too variable from unit to unit, almost impossible to quality control within ±0.1 dB. Driving the tweeter, the midrange, and the woofers directly, with three separate amplifier channels, is a much purer solution. The full damping capability of each amplifier is utilized that way. Dr. Fortier is an electronic crossover specialist, and the crossover/control unit he specified for the Mach 17 is unique. The circuit implementation is by Bryston, with pretty much the same topology as the classic Bryston 10B. (Bryston actually manufactures the unit for Waveform.) The uniqueness of the design is in the control functions, as I'll explain in a moment. The crossover frequencies are 325 Hz and 1850 Hz; the filters are of the fourth-order Linkwitz-Riley configuration with 24 dB per octave slopes; the gain for each of the three channels per side is internally set to match the efficiency of the corresponding driver—all very pure but not adjustable by the user. The adjustable controls are for (1) midbass level, centering on about 90 Hz, with a range of 6 dB; (2) a very subtle midrange "presence" boost, up to 2 dB; and (3) treble attenuation, 4 dB or less. These are extremely sophisticated controls, almost comparable to those on a Cello Palette (hey, I said *almost),* and they serve the same purpose: to touch up the frequency balance of recordings that do not necessarily sound best when played back flat in a particular listening room. Since the Mach 17 is an extremely accurate loudspeaker without a sonic signature of its own, every recording played through it sounds a little different—that's only logical—and may or may not need a slight correction unique onto itself. Waveform supplies tiny stickers to mark each CD with the preferred bass/midrange/treble setting.

Six power amplifier channels with exactly the same gain are needed run a pair of the Waveform speakers. The electronic crossover unit provides no compensation for amplifiers with different gains. Furthermore, it has balanced inputs and outputs only. Thus it is somewhat unlikely that the potential purchaser already has all the required electronics. I used a three-channel Bryston 5B ST (120 watts per channel) on each speaker, with the balanced output of my trusty old Boulder MS preamplifier feeding the electronic crossover. (Many a high-endnik would call that a medium-priced system.) I also tried the Mcintosh MC7106 six-channel power amplifier, using XLR-jack-to-RCA-plug interconnects, with equal results.

John Ötvös is rather proud of the fact that he and Dr. Fortier did not even listen to the laboratory prototype of the Mach 17 until the measurements appeared to be incapable of further improvement. That coincides very much with my own audio philosophy—measurable, i.e., provable, accuracy is where it's at. (In your face, Bob Harley.) Thus the measurements tell the whole story—and it's quite a story. Now, the audio measurement facilities of the National Research Council in Ottawa, Canada, where the speaker was developed, are among the finest in the world (not to mention their blind listening setup and protocols). The laboratory of ***The Audio Critic*** is very good but not *that* good. Among other things, we have no anechoic chamber. Thus, I tend to trust the official measurement printouts of the NRC even more than I trust my own. I verified the NRC test results as conscientiously as I knew how, using the MLS capability of the Audio Precision System One (which yields quasi-anechoic curves), as well as all the other test methods I have previously described in my loudspeaker reviews. My measurements tracked those of the NRC quite closely, just as I expected on the basis of past experience, so I dispensed with the grain of salt I always keep handy for manufacturers' specs.

The frequency response of the Mach 17 is almost as flat as that of an amplifier. I have never seen flatter response on axis. The NRC curve shows ±1 dB from below 30 Hz up to 20 kHz. My own curve departs from that by less than 0.5 dB. I'm willing to buy the NRC version. The off-axis response is equally remarkable. The midrange doesn't change at all at 15° and 30° off axis horizontally. The tweeter holds up equally well up to 11 kHz, then rolls off 12 dB per octave at 30°. (The Snell Type A has the same tweeter, so I was already familiar with this type of response.) At 60° off axis the midrange barely budges and the tweeter still holds nicely up to 7 kHz, above which it falls precipitously. This wide-dispersion characteristic is a very important part of the design, as it determines the power response into the room. Vertical movement of the measuring microphone results in a notch or trough at the 1.85 kHz crossover as one leaves the ideal measurement height, but that is to be expected. The effect is more obvious when the microphone is too low (i.e., below ear height when seated) than when it is high (i.e., ear height when standing). The design intends to accommodate both seated and standing listeners. (I happen to stand as much as I sit when listening.) For some reason, the Waveform/NRC literature does not deal with the variability of response in the vertical plane, although the speaker stands up very well in that respect.

As for the bass response, I am more comfortable with my own nearfield measurements than with the NRC's anechoic ones, in which I suspect the chamber size dominates the true inflection point and rolloff characteristic, at least to some degree. I found the bass to be

±1 dB from 60 Hz down to 21 Hz when I summed the two woofers and two vents at a certain nearfield point (which may or may not have been the most representative point), with the midbass control at the 0 setting and midrange at maximum. A 6 dB per octave low-frequency boost is always in the circuit from about 90 Hz down; that's part of the design for flat bass. That plus the two-woofer/two-vent configuration and the variable midbass level make it a little more complicated than usual to measure the true response, but I also took a few in-room farfield measurements and could see that the response is never down more than 2 or 3 dB at 20 Hz. That's not quite like having a super subwoofer in the system, but it's about good as it gets with a monolithic speaker. If you want totally flat response down to the bottom limits of audibility, John Ötvös is not going to sue you if you add a big Velodyne or Bag End to the Mach 17, but at that point the price tag is beginning look less amazing, and the musical pleasure increases only minimally. (Yes, there *is* a small difference, but only on music containing a great deal of energy at the lowest frequencies. And even that perception may be due to my surreptitious habit of setting my subwoofers for slightly tipped-up response below 30 Hz—I'm addicted to those lowest of the lows. Don't tell anybody.)

What about distortion? Here I must distinguish between the NRC method of measurement and my own. Regardless of the SPL at 1 meter or 2 meters, I must measure the THD + N right off the driver diaphragms because at 1 meter or 2 meters the background N of the room totally swamps the measurement. That is not the case in the NRC's dead-quiet anechoic chamber. They are able to measure the distortion at the same location as the frequency response and SPL. I think that's preferable, so I tentatively accepted the NRC data and then followed up with my own measurements to make sure there were no gross differences. There weren't any, but remember that what matters in this case is the general trend of the numbers, not some kind of Bureau of Standards level of accuracy.

Overall, the distortion of the Mach 17 stays below 0.8% from 70 Hz up at 85 dB SPL at 2 meters and below 0.6% from 70 Hz up at 91 dB SPL at 2 meters. (The 1-meter SPL equivalents are 91 dB and 97 dB, repectively.) At the higher level there is a packet of distortion centering on 4.5 kHz and peaking at 2.5%, but the tweeter is quite unlikely to be driven that hard at that frequency. These figures are very respectable but not quite Velodyne DF-661-like. They could be lower. At most frequencies and SPLs that will probably make no difference, but at the ear's maximum sensitivity to second-harmonic distortion, in the neighborhood of 2 kHz, it could be audible. There exist various divergent opinions and estimates as to the ear's thresholds, so I tend to favor the Velodyne approach just to be on the safe side, but that doesn't mean the difference between low distortion (Waveform) and *very* low distortion (Velodyne) is all that obvious.

Below 70 Hz, the Mach 17's bass sytem is no lower in distortion than other good woofers, rising to 3% in the 30 to 40 Hz band at the lower SPL cited above and to 6% at the higher SPL. That includes the 6 dB per octave low-frequency boost which is always in the circuit. It is my understanding that ultralow-distortion motional-feedback woofers from Velodyne were considered for the system and would have been available on an OEM basis, but their 100 Hz upper limit just did not fit the system architecture, not to mention that they would have required dedicated power amps.

Now I'm ready to tell you more about the sound. In some ways it's my favorite sound since I started testing loudspeakers. It's not tall and wide and reverberant, being essentially a point source, as I already said. It's just dead accurate and utterly transparent. It has a *thereness* that you immediately know is right. Somebody said, "like a giant pair of headphones," which is true as far as the absolute immediacy of the sound is concerned but not true insofar as headphones are a pain in the tushy and tend to jam the soundstage into the back of your skull. *Au contraire*, the Mach 17 presents a gorgeous open soundstage with marvelous localization. Not only that, but it gives you that relaxed, reassuring feeling that all is well, exactly as things should be. Some listeners have told me that the speaker is a bit lacking in upper bass, but in my opinion they were merely disagreeing with my setting of the controls on certain CDs. Ideally one should reset the controls for best balance on each recording and note the settings on one of those bass/midrange/treble stickers supplied with the speaker, as I already explained. I'm too lazy and insufficiently geeky for that. I leave the controls in a generally satisfactory all-purpose position, from which I depart infrequently. If the Mach 17 is lacking in anything, it is the lowest bass, not the upper—but I have already dealt with that. There may also be excessive brightness or edgines if the room isn't dead enough. The extremely wide dispersion of the speaker involves the reflective surfaces of the room much more than you may be used to. I deployed sound-absorbent screens near the sidewalls close to the speaker and was very happy with the results—not a trace of edginess, just delicate and airy highs. The treble control has a relatively narrow range and should be used only to compensate for the recording, not the room. In general, "people who live in glass houses" should think about extensive room treatment if they want a Mach 17.

Readers of the last issue know that my favorite loudspeaker system until then was the Snell Acoustics Type A. Since the entire Ötvös-Fortier effort started, evolved, and was finalized at the NRC facility, it seemed only natural that I should do a listening comparison of the Mach 17 against the Type A in accordance with the NRC protocol, as suggested by Floyd E. Toole. That means mono versus mono, *mano a mano,* side by side, in the middle of the room, at matched levels. That way the seductions and distractions of stereo do not intrude; it's the strictest possible test of tonal quality. At the NRC they do it double-blind, but I have no acoustically transparent black screen as they do. I did match the SPLs as accurately as I could, however (with a sound level meter, not by ear). Well, what do you know, the two sounded much more alike than I expected, but then again both have the same tweeter, and the Snell is also based on the NRC philosophy, so maybe I shouldn't have been surprised. I thought that the Snell with its spread-out driver complement "editorialized" about its input a wee bit more than the Waveform, which just rendered a crystal clear replica of the signal right in your face; the difference, however, wasn't at all dramatic and might have been even smaller if more room had been available to listen from further back. I'll take either speaker, and at one third the price of the Snell the Waveform is of course the more attractive buy. Indeed, a steal. (That doesn't mean the vested interests of the Ultrahigh End, including the usual suspects in the tweako reviewer community, will not be tempted to badmouth the Mach 17, as it is a threat to speakers in the upper five and six figures. What I'm telling you is that you can safely ignore such reviews.)

My bottom line recommendation: Instead of being a sucker and spending thousands of dollars on a high-end preamp or other shiny new toy that can't possibly change your audio life, call John Ötvös and start negotiating for a Waveform Mach 17. It will cost you less than a Krell CD player, even at the 1997 price, and it *will* change your audio life. I guar*awn*tee you (as the Cajun TV cook Justin Wilson would say).

❖ ❖ ❖

Transducer Miscellanea by David Rich

ACI Sapphire III

Audio Concepts, Inc., 901 South 4th Street, La Crosse, WI 54601. Voice: (608) 784-4570. Fax: (608) 784-6367. Sapphire III compact 2-way loudspeaker system, $999.00 the pair (direct from ACI). Tested samples owned by reviewer.

This is the latest version of the continuing revisions of the Audio Concepts (now called ACI) Sapphire series. When we tested the previous version, the Sapphire II*ti*, we found it to be a step backward because of a change of tweeter. It sounded bright and edgy. ACI traded my II*ti*'s for a pair of cosmetically damaged IIIs. The damage must have been visible only with a tunneling microscope because they looked perfect to me. It was a good deal for me because the II*ti*'s were sitting in the garage waiting to be sold. The IIIs are back in one of my listening rooms and they are going to stay right there.

In the Sapphire III the tweeter is changed again, this time to a simple soft dome found in many top-of-the-line speakers preferred by the high end community. Our measurements show that the performance of the speaker is similar to that of pervious incarnations except that the tweeter level has been reduced by about 2 dB. Since it was flat before, this seems like a misguided attempt to make the speaker sound less bright. I think it was the resonances of the titanium Focal tweeter that caused me to find the II*ti* so unpleasant.

The sonic result is a speaker that sounds very similar to the what I remember the original Sapphire II sounded like. Since the speaker continues to have first-order crossovers, placement is very critical. Big suckouts in the frequency response occur is you are not on the correct woofer axis. When set up correctly, the speaker sounds good but it does not have the transparency of the Monitor Audio Studio 6. The Monitor Audio is more forward than the ACI, and that might tip the balance toward the ACI for some, but to me the Monitor Audio is the clear winner. The comparison between these two speakers used to be very unfair, but ACI has meanwhile raised the price of the Sapphire III to $999.00 the pair, factory-direct before shipping charges (shipping is free at the moment), whereas the Monitor Audio has actually dropped in price to $1999.00 the pair—but that is the full retail price and discounts from list are a reality. Add to this the fact that you have to pay reverse shipping charges if you do not like the Sapphire, and we see this is not as unfair a comparison as it used to be.

Comparison with the JosephAudio RM7si brings things closer in price. (The RM7si lists for $1299.00 the pair.) I am somewhat more impressed by the quality of the ACI box, but in crossover design the comparison clearly favors the Joseph. The Joseph's bass tuning makes it a better stand-alone speaker, but the ACI would have the edge in a satellite/subwoofer system. The better crossover of the Joseph resulted in a much more balanced sound quality, with much less finicky setup, and its flatter frequency response yields more definition and detail. On the other hand, in my room the ACI speakers produced a warmer, more spacious sound than the Josephs. (Not in the larger room of the Editor, however.) The reasons for this may have been quite mundane, such as for example the slanted baffle of the Sapphire III, which adds to the reflected energy in the room. Also, some of the frequency-response effects associated with first-order crossovers may have entered the picture. I found the Spica TC-50 and Vandersteen 2C to create a similar warm, spacious effect.

If you held a gun to my head and said "Pick!"—I would cop out. For a bedroom setup with no subwoofer, go with the Joseph. Optimal placement required by the Sapphire is not going to happen under these conditions, and the Joseph will have more bass. If you have the room to place them optimally and have a sense of adventure (or a low tolerance for frustration in case you do not like them and have to send them back), you might want to try the Sapphire III + Titan (the powered version of the Sub 1 we tested long ago). Do not confuse this with a mix-and-match satellite/subwoofer system. The manufacturer has designed these two units to work together so you will not have to play speaker designer, and that is a very, very big plus. I have never heard a satellite/subwoofer system work correctly that was not designed from the ground up to do that function. Crossing over a three-piece system at 100 Hz is not something that can be done without very careful design and integration. When a skilled designer does it (the Spica TC-50 plus Spica Servo subwoofer for example), it can work very well. Roll your own and I guarantee trouble.

So for a total price of $1800 ACI will sell you a full-range high-end system that is phase-coherent (so the tweaks will respect you), has the "boxless" sound quality of a satellite/subwoofer system, and is capable of playing with full dynamics. Other priorities would lead to other choices. While I did not have the Titan, I did try the Sapphire III with the Sunfire subwoofer. No way would I trade that setup for my Sound Lab Quantums. For me, 20 Hz bass and big dynamics just are not as important as transparency and freedom from colorations.

The ACI Sapphire III is still a very well-constructed product made with quality drivers and a rock-solid cabinet. Its relative lack of transparency probably has to do with the low-order crossover. This causes the drivers to work very hard outside their passband. It also causes the polar response of the speaker to be poor, and that must affect the farfield response. I suggest that ACI try a high-order crossover for the Sapphire IV.

Grado Laboratories SR125

Grado Laboratories, Inc., 4614 7th Avenue, Brooklyn, NY 11220. Voice: (718) 435-5340. Fax: (718) 633-6941. The Prestige Series SRI25 dynamic headphones, $150.00. Tested sample on loan from manufacturer.

We do not have our objective measurement setup for headphones in place yet, but I did not want to wait to tell you about these nice headphones. John Grado would tell you they do most of the design on a subjective basis, so he would not be concerned about objective test results anyway. I suspect that the frequency response of these phones may not be perfectly flat because of this (the Grado cartridges, designed by John's uncle, never challenged Shure in this respect), so I would not use them as a monitor during recordings. But whatever John Grado has done works very well when you use these phones to listen to CDs in a home setting. They sound very transparent. This is especially noticeable in comparison with the headphones that come with portable CD players. Ambience is well reproduced on the SRI25. The top end is very clean, with none of the burn-your-ears-off quality found on many headphones, but the highs by no means sound rolled off. Low end is subjectively flat to about 50 Hz; 40 Hz is still quite audible but 30 Hz is gone. Cheap phones are often bumped up in the midbass and then the response dives out of sight.

You can listen to these headphones for hours, both because they are comfortable to wear and because the sound is clean and subjectively flat enough not to introduce listener fatigue. They are sensitive enough to be used with portable equipment. My only complaint is the thick cord used by Grado, which makes the phones heavier and more clumsy than they have to be. This is probably some tweako this-cable-sounds-better move.

At $150 I highly recommend you give these a listen. You might also want to look into the SR60 at $69. It has a different voice-coil design and diaphragm, as well as a less expensive headband spring.

* * *

Editor's Note: *Measuring headphones accurately requires a dummy head with microphone ears, something we are definitely not getting in the near future. The application is too narrow for the high cost of such a device. I did measure the SR125, however, in my own crude way, just jamming a microphone into it in various open and blocked positions. That will indicate a general trend without yielding accurate numbers. The phones appear to be more or less flat from about 2 kHz down to the bass region, where a gentle rolloff begins at approximately 90 Hz or 80 Hz or a little lower, depending on the tightness of the seal. There is a weird and very steep rise from 3 kHz to 4 kHz, above which another gentle rolloff begins. There is still considerable response at 14 kHz but not much at 20 kHz. That's how vague I'll have to leave it. These are clearly not superflat headphones.*

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Catching Up on Sophisticated Audio Electronics, Analog and Digital

By Peter Aczel
Editor and Publisher

Although no well-designed electronic signal path for audio has a sonic signature of its own (down, tweaks, down!), there are still many reasons to prefer one particular design over another.

I am sure we have some new readers to whom it is still a strange new idea that any correctly engineered piece of purely electronic (viz., not electroacoustic) audio equipment, be it analog or digital, is indistinguishable in sound from any other when each is used within its intended output capability. Such readers will look for fulsome descriptions of the upper midrange, front-to-back depth, imaging, etc., of the equipment reviewed below and won't find any. I recommend back issues of this journal, all the way back to No. 16, as a rehab course for these brainwashed audiophiles. (See also the "Paste This in Your Hat!" piece on page 9 of this issue.) Sorry, all you longtime readers, for constantly repeating myself on this subject, but the tweako magazines also repeat themselves and new ones keep coming out of the woodwork.

Compact Disc Player

Accuphase DP-55

Accuphase Laboratory, Inc., Yokohama, Japan, through Axiss Distribution, Inc., 17800 South Main Street, Suite 109, Gardena, CA 90248. Voice: (310) 329-0187. Fax: (310) 329-0189. DP-55 compact disc player with RC-18 remote control, $3995.00. Tested sample on loan from distributor.

This is Accuphase's new bottom-of-the-line CD player. Whaaaat? Bottom of the line at almost $4000? That's right. Their top-of-the-line CD player lists at $7495.00. (Accuphase also has a $16,495.00 preamplifier, and other such goodies.) Now, what does that tell you? Yes, limited production, obviously, and a big chunk for the distributor, inevitably (not to mention an unfavorable yen exchange rate)—but what else? I'd say they want to sell a few CD players to audiophiles who are merely well-to-do and a little crazy, instead of super rich and stark raving mad. They want the Accuphase cachet to rub off on "modestly priced" equipment, if you'll pardon the expression.

Well, the DP-55 has gorgeous metalwork, cosmetically equaling the impact of the top-priced Accuphase models, but it's another story when you remove the cover and look inside. There the quality is on the level of a standard brand-name Japanese product. Single-sided PC boards are stuffed with good commercial-grade parts. Nothing wrong with that—everything is neatly constructed and nicely laid out, but it isn't "electronic jewelry" like the highest-priced Accuphase equipment. We expected a little more for $4K.

In terms of circuit design, however, there are some quality touches. Preeminent among these is the use of three paralleled DACs per channel. (The higher-priced Accuphase CD players use even more, up to 16 per channel.) Theoretically, this should reduce the noise floor, but there's a catch. The DP-55 uses the 20-bit Burr-Brown PCM 1702 surface-mounted DACs—without any grade markings. That seems to indicate they are not the highest (K) grade, raising the question whether a single K-grade PCM 1702 per channel would not be actually preferable, as regards both noise floor and linearity. Furthermore, the surface-mounted DACs appear to be hand-soldered (low tech) rather than wave-soldered (high tech). The digital filter is the NPC SM5843AP, not quite the best. The transport appears to be a plain-vanilla Sony OEM unit, without linear tracking. A quality feature, on the other hand, is the processor mode—a front-panel button turns the DP-55 into an outboard D/A converter with inputs, coaxial and optical, for digital signal sources such as a DAT recorder, MD recorder, etc. There are also the usual digital outputs, coaxial and optical, in addition to the analog outputs, both balanced (XLR) and unbalanced (RCA). At $1800 it would impress the hell out of me.

On the lab bench the measurement results were excellent but didn't break any records. In the 16-bit CD playback mode, full-scale THD + N across the audio spectrum was the usual (for high-end equipment, that is) 4 to 4.5 dB short of the theoretical limit of-98.08 dB , as against only 1 to 2 dB in the case of Sony's best (for less money). Once again this proved to be gain-related analog distortion; on a -24 dB track the excess distortion went down to virtually zero. Most other test-CD measurements were close to perfection. Quantization noise: - 97.8 dB in the better channel, -96.5 dB in the other. Dynamic range: 97.7 dB. Channel separation: 118 dB to 128 dB, depending on frequency and the channel measured. Spectrum of 1 kHz tone at -90 dB with dither: no trace of harmonics. Just a little less perfect were the low-level linearity error, about -0.3 dB at the -90 dB level and below, and the monotonicity pattern, which at a couple of points showed insufficient differentiation between adjacent LSB steps. (K grade would have prevented both of these minor boo-boos.)

In the processor mode, with a 16-bit input, the above results remained the same—not surprisingly. I did try the by-now-trivial "Rob Watts test" (FFT spectrum of a dithered 1 kHz tone at -60 dB) and obtained a plot with very small second-, third-, and seventh-harmonic glitches rising out of a -128 dB noise floor, in contrast to some squeaky-clean plots and -130 dB noise floors in a few past instances. Increasing the input word length to 20 bits (and/or the sampling rate to 48 kHz) produced only very slightly improved results. Excess distortion at full scale got better by less than 2 dB; the irreducible noise floor went down by just a very few hard-to-read dB; that's about it.

One more thing. I generally don't test the transport mechanism of a high-end CD player for error correction because in present-day units it's not an issue, but in this case I did, in view of my remark above about the OEM transport. Dropouts of 0.75 millimeter or smaller were handled without a glitch; dropouts of 1.00 millimeter or larger were troublemakers. I would call that good but not exceptional performance.

Overall, I see little or no temptation for an audiophile to spend $3995 on this unit. The last three top-of-the-line Sony ES players I tested were all slightly better performers, less costly, with more advanced electronics and mechanics.

Headphone Amplifier

Audio Alchemy HPA vl.O

Audio Alchemy, Inc., 31133 Via Colinas, Suite 111, Westlake Village, CA 91362. Voice: (818) 707-8504. Fax: (818) 707-2610. HPA vl.0 headphone amplifier, $259.00. Tested sample on loan from manufacturer.

Audio Alchemy appears to have assumed a very low profile lately in the audio community, for whatever reason. All I know is that they sent me this unsolicited review sample some time ago, and I happen to like it sufficiently to report on it here.

This is a modem-sized unit with line-level inputs and an outboard ±14 V power supply. It accepts a single pair of headphones via a standard stereo phone jack, and it also incorporates the HeadRoom "audio image processor," designed to make headphones sound less "headphoney." (But who held a gun to your head to make you listen to headphones if you don't like their sound?)

The HPA vl.0 is definitely a low-distortion device. Into 150Ω, which is reasonably representative of a headphone load, minimum THD + N is -90 dB at 20 Hz and 1 kHz (6 V out), and -80 dB at 20 kHz (3 V out). That indicates a small amount of dynamic distortion, not enough to worry about. The distortion is completely noise-dominated. Switching on the HeadRoom processor raises the 20 Hz and 1 kHz distortion by 5 dB but leaves the 20 kHz distortion unchanged. What does the processor actually do? It inserts a shallow S-shaped equalization curve (maximum boost 2.6 dB at 9 kHz) into both channels, introduces frequency-dependent phase differences between left and right, and reduces channel separation drastically. With the processor off, channel separation is 42 dB at 20 kHz and increases by 6 dB per octave with decreasing frequency, until it reaches 83 dB at 20 Hz. With the processor on, channel separation is 10 dB at all frequencies below 1.5 kHz and increases slightly at the higher frequencies, until it reaches 20 dB at 20 kHz.

The sound of the HPA vl.0 is of course perfectly clean and neutral with the processor off. With the processor on, it sounds—well, different. I don't feel strongly about it one way or the other. I happen to believe that only true binaural recordings, recorded with a dummy head, are fully compatible with headphones. Standard stereo recordings, intended to be heard through loudspeakers, all sound a little weird through headphones, with or without processing. I'll leave it at that. If your stereo system does not have a convenient headphone output, the Audio Alchemy unit is a well-engineered solution worthy of recommendation.

Power Amplifier

Bryston 8B ST and 5B ST

Bryston Ltd., P.O. Box 2170, 677 Neal Drive, Peterborough, Ont., Canada K9J 7Y4. Voice: (705) 742-5325. Fax: (705) 742-0882. 8B ST 4-channel power amplifier, $2995.00. 5B ST 3-channel power amplifier, $2465.00. Tested samples on loan from manufacturer.

Regular readers of this journal know that a Bryston power amp is always a completely predictable performer. There are very few audio equipment brands about which I can say they are as good as money in the bank, but Bryston is one of them. Chris Russell's basic amplifier circuit concept, frequently discussed in our pages, has changed very little over the years; in this instance it has undergone some refinements. Chris's engineering associate Stuart Taylor (ST) is recognized as a layout guru, and he has come up with entirely new physical layouts for these multichannel amplifiers in order to simplify the signal paths and bring distortion, hum, and noise down to new low levels. Another improvement has to do with what Bryston calls their input-buffer-with-gain, also designed to lower distortion and noise, but certainly not a major change. In all essentials, a Bryston is a Bryston is a Bryston—just specify how many channels you want on one chassis and how much power per channel. Whatever configuration you choose, you'll have an amplifier on the leading edge of the art.

The rated power per channel of the 4-channel 8B ST and 3-channel 5B ST is the same: 120 watts into 8Ω. My measurements showed 150 watts to be available at extremely low distortion: between -96 and -98 dB (barely over 0.001%) at any audio frequency. Into 4Ω the output did not quite double but reached 225 watts with -90 to -96 dB distortion, the least good figure being the 20 kHz reading (-90 dB = 0.003%). The THD + N curves were entirely noise-dominated and indicated extremely low noise even at only 10 milliwatts output (-61 to -64 dB, depending on the load). With two channels bridged, I measured 400 watts into 8Ω with distortion that dipped as low as -100 dB at 1 kHz and -93 dB at 20 kHz.

The PowerCube test (see Issue No. 20, pp. 16-17, for a complete explanation and illustrations) painted a pretty decent picture, with 39.5 V into 8Ω/0° (195 W), quite gently declining voltage into all the purely resistive loads down to 1Ω, and always slightly higher voltage into the reactive loads than into pure R. Into 1Ω/0° there was still 22.3 V (497 W) available. (Remember—the PowerCube uses 1 kHz bursts of 20 ms duration, limiting the amplitude at 1% distortion.) For better performance you would have to go to one of the mega-power-supply amplifiers. With two channels bridged, the PowerCube slopes much more steeply, since 2Ω and 1Ω loads are not a good match to the bridged output stages, but there are still no anomalies into reactive loads. Into 8Ω/0° the PowerCube reading was 72.5 V (657 W) in the bridged mode.

Needless to say, the frequency response of each Bryston channel is dead flat. At approximately 1 watt into 8Ω I measured 0.0 dB deviation from 10 Hz to 2 kHz, -0.04 dB at 20 kHz, and -0.22 dB at 50 kHz. Channel separation as measured at the same output level is OK but not great: 40 dB at 20 kHz, increasing by 6 dB per octave at decreasing frequencies, reaching 64 dB at 1 kHz and 90 dB at 20 Hz.

The obvious comparison that comes to mind here is with the Mcintosh MC7106, which is a 6-channel model listed at $3500.00. The Bryston amps do quite a bit better on the PowerCube, indicating a power supply advantage, but the Mcintosh is even lower in distortion (though not at 20 kHz in the bridged mode) and slightly higher in power output before clipping. The noise floor is a very close contest, but in channel separation the Mcintosh wins. The Brystons both have balanced (XLR) inputs in addition to the standard RCA phono jacks; the Mcintosh does not. On the other hand, the Mcintosh is dead silent mechanically and electrically, whereas both Brystons have a slight mechanical hum coming directly from the power transformer, especially when cold, and produce small but audible on/off thumps through the speakers. As for warranties/guarantees, Bryston is way ahead with their free and unconditional 20-year deal, but the Mcintosh amp offers more complete electrical protection against failure and abuse. On balance, I'd say the audio purist will lean toward the Brystons and the convenience seeker toward the Mcintosh, but overall it's a win-win situation.

In any event, I rate both the Bryston 8B ST and the Bryston 5B ST in the tip-top category of power amps.

Power Amplifier

Carver Research Lightstar Reference

Carver Corporation, P.O. Box 1237, Lynnwood, WA 98046-1237. Voice: (206) 775-1202. Fax: (206) 778-9453. Lightstar Reference dual-monaural power amplifier, $3995.00. Tested sample on loan from manufacturer.

This is the amplifier Bob Carver *almost* finished before he bowed out of Carver Corporation and started his new Sunfire enterprise. His old engineering department finished the job but left the basic design alone. As a result the Lightstar and Sunfire amplifiers are virtually identical. Both are very conservatively spec'd at 300/600 watts per channel into 8Ω/4Ω. David Rich went over both circuit schematics with great care and reported only minuscule differences. (See his analysis of the Sunfire circuit in Issue No. 22, pp. 31 and 49, and my report on the Sunfire amplifier's measurements in Issue No. 23, pp. 25-26. I shall try not to be repetitious here where the same comments would apply.) The differences worth noting are the two entirely separate power supplies of the "dual-monaural" Lightstar (with two separate line cords for emphasis), as against the shared power supply of the Sunfire, and the much more luxurious sculptured metalwork of the Lightstar Reference, for which you pay plenty. (Carver Corporation also offers a stripped-down Lightstar for $1500.00 less.)

Since two separate power supplies and a gorgeous chassis are certainly not undesirable, I would prefer the Lightstar Reference as a birthday present, but with my own money I would be more likely to buy the $2175.00 Sunfire. In general, it would appear that Bob Carver is more intense about maintaining a performance-per-dollar competitive edge, and his Sunfire product shows it. The

Lightstar Reference clearly wins the beauty contest, however, as well as the techie showoff match.

As for measurements, you will recall that the Sunfire didn't exactly shine (no pun intended) when it came to THD + N. Interestingly enough, the Lightstar Reference measured at least 10 dB better at nearly all frequencies under nearly all test conditions, and in a few instances a lot better than that. At the lower frequencies the Lightstar is almost a "normal" high-end amplifier. Would you believe 20 Hz distortion of -96 dB (into 8Ω before the onset of clipping) with a 22 kHz filter, rising to -78 dB with an 80 kHz filter? The Sunfire falls 15 dB short of that. On the other hand, there is a convergence toward higher distortion in both amplifiers at the higher frequencies. At 6.5 kHz the Lightstar is still ahead by a few dB, but the 0-to-80-kHz FFT spectrum of a 19.5 kHz test signal at 200 W into 8Ω looks pretty much the same through both amplifiers. (The Lightstar is actually still a little cleaner up to 40 kHz but not into the next octave.) The circuit schematics offer no explanation for these basically not very important differences.

The PowerCube test also appeared to indicate that the Lighstar Reference is virtually identical to the Sunfire. Whatever minor differences I recorded were due to the slightly different test setups for the two amplifiers, and I don't want to trivialize the basic issues here by nitpicking those differences. Both amplifiers are very happy with loads of 8Ω, 4Ω, and 2Ω, regardless of phase angle, and neither amplifier is able to maintain its top voltage output into 1Ω loads, resisitive or reactive, with less than 1% distortion—for whatever reason. Even so, I cannot imagine any real-world loudspeaker load that either amplifier couldn't drive with aplomb. Just for the record, the PowerCube reading into 8Ω/0° was 55.1 V (380 W) for the Lightstar Reference.

Channel separation, i.e., crosstalk, is another valid point of comparison. The Lightstar is not quite as good in this respect as the Sunfire but almost. At the lower frequencies separation is of the order of 70 dB; in the vicinity of 1 kHz it is 66 to 68 dB; above that the left channel leaks into the right much more than vice versa. Worst case: 44 dB at 20 kHz, but the other channel never gets worse than 57 dB. I have no serious problem with that.

One thing the Carver engineers did differently after Bob had left was to leave out the 1Ω series resistor that makes one pair of terminals a "current source" on the Sunfire amp. The Lightstar has two pairs of paralleled terminals per channel, all low-output-impedance voltage sources. If you've read my Sunfire review, you know I'm not going to fault the Lightstar on that count. In fact I have no reason to fault it at all, once I have accepted the unorthodox design principle, except maybe on price. I still don't know what will become of "Hamlet without the Prince of Denmark," viz. the Carver Corporation without Bob Carver, but this power amplifier is a fine product.

Third-Generation MiniDisc Recorder

Sony MDS-JA3ES

Sony Electronics, Inc., 1 Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 930-4748. MDS-JA3ES MiniDisc deck with RM-D2M remote control, $1200.00. Tested sample on loan from manufacturer.

The MiniDisc is now about four years old and hasn't really taken off yet. Nor is it dead. I understand they love it in Japan and that's what seems to be keeping it alive. I must say I love it too, at least as a physical format. It's small, cute, and cuddly, as well as ruggged and practical, and also delightful to edit and track-access. If the digital era had been launched in this format, or better yet, if the digital era had come earlier and this format had emerged instead of the Philips cassette, we would all be better off today. (Of course, I'm also assuming that some kind of incredibly early engineering wizardry—blue lasers, etc.—would have made the 5-to-l data compression unnecessary, so that the audibility/inaudibility of perceptual coding wouldn't be an issue. Hey, I'm allowed to dream...)

The trouble with MiniDisc as a recording/playback medium for consumers is that after four years a blank disc still costs about six times as much as a top-quality blank audio cassette. That's outrageous and simply not competitive, no matter how much more attractive the MiniDisc may be. Sony has insisted from the beginning that the MiniDisc is an improved successor to the Philips cassette, not the CD, but under the circumstances it is neither. It is merely a promise and a rather costly techie toy. Lots of fun, though.

In Issue No. 21, I reviewed in some detail Sony's second-generation MiniDisc deck, the MDS-501, and presented my views on the then fairly new MD format, the ATRAC compression algorithm, and the procedure to test an MD deck. I see no reason to cover the same ground all over again in this review (No. 21, which also contains Bob Adams's timeless tutorial on digital jitter, is still available to our readers at the prorated subscription price), so the following is basically an update rather than a complete MD treatise.

The JA3ES is the ultimate MD deck, the high-end embodiment of what was originally conceived to be a mass-market technology. It incorporates third-generation ATRAC, which is allegedly more nearly transparent in sound than its predecessors, and it costs $200 more than the already pricey MDS-501 it replaces. It is without question an impressive piece of gear; what I wrote about the look-and-feel and various features of the MDS-501 goes in spades for the JA3ES. Sony took the ES ("Elevated Standard") suffix seriously when they added it to the model number.

Just as an example, the highly touted CXD8504M digital filter chip used in Sony's $3000 top-of-the-line

CD player also appears in the JA3ES. (Don't look for a David Rich autopsy of the circuit boards, however. I wouldn't dare send him something as crass as a perceptually coded digital device.)

As I explained in my MDS-501 review, I don't see much point in measuring what the ATRAC circuitry does, since the performance criteria are psychoacoustic and have little to do with I/O accuracy. I suppose one could measure adherence to the specified codec protocol, but I didn't think that was worth the trouble. I did perform, however, the usual analog and digital accuracy tests. The JA3ES handily surpassed the second-generation product in these straight-through measurements.

Both the A/D and D/A converters are extremely linear; the delta-sigma DAC has ±0.2 dB gain-linearity error down to -110 dB (!) and the ADC is off by only +0.6 dB at -90 dB. Full-scale THD + N is in the -92 to -93 dB range over most of the audio spectrum with a 16-bit input (improving by only 3 to 5 dB with a 20-bit input, although the front panel advertises 20 bits). Reducing the digital input from 0 dB to -20 dB results in only a 1 dB improvement in excess THD + N above the theoretical minimum, eliminating gain-related analog distortion as the source. Indeed, the purely analog line-in/line-out distortion of the JA3ES is of the order of -85 to -89 dB at all frequencies, right up there with the best. Here's one for the book: the Rob Watts D/A test (FFT spectrum of a dithered 1 kHz tone at -60 dB) shows the JA3ES to be just a little cleaner than the $3995.00 Accuphase DP-55 CD player! There is no glitch whatsoever rising from the -126 dB noise floor.

As for the subjective transparency of the perceptual coding used in the JA3ES, I did not set up in-depth ABX comparisons with listening panels because, as I said, Sony claims no parity for the MD with CD, DAT, or any other uncompressed digital or analog medium. I did, however, try to confound the ATRAC with tricks I had learned at the October 1995 convention of the AES in New York. (There was a professional workshop there called "Listening Test Standards for Evaluation of Low Bit-Rate Codecs," in which all data-reduction methods were asserted to be in need of improvement and "killer" signals were demonstrated.) I must say that I was unable to trip up the Sony even with evil intentions. That doesn't mean it can't be done, but I just couldn't find the signals to do it with. My admittedly limited listening tests have revealed no obvious obstruction of transparency in third-generation ATRAC. This statement is based on digital-to-digital copying of good CDs with the JA3ES, not on comparing any prerecorded MD release with the CD version.

The long and the short of it is that I like the MD a lot, certainly at the JA3ES level of performance, and wish its survival could be assured by lower blank-disc prices and generally better U.S. marketing. Failing that, the Betamax scenario is inevitable.

Compact Disc Player

Thorens TCD 2000

Thorens of America, Ltd., 84-05 Cuthbert Road, Kew Gardens, NY 11415. Voice: (718) 847-4289. Fax: (718) 849-7698. TCD 2000 compact disc player with TFB 2200 remote control, $2500.00. Tested sample on loan from manufacturer.

Nearly every graying audiophile like me has owned a Thorens turntable at one time or another in the course of the phono era. Thorens apparently believes that the turntable image will sell CD players because this unit is made to look like a miniature but very high-tech phono turntable. The platter, fully visible at all times under the plexiglass lid, has a gold mirror finish and quivers like jelly on a soft suspension that must be screwed down for transit. What's more, the whole black-and-gold affair is about half the size of a normal CD player, matching a complete line of cosmetically coordinated Thorens mini-separates. Is it an appealing look? Yes, in a sort of Euro-trashy way. I'd be more impressed if the line cord were standard U.S. instead of just-off-the-boat Continental with an add-on adapter.

The overall system design, other than the transport, appears to be Philips-based, with Bitstream conversion and somewhat rudimentary control functions. I say that because my *bête noire* when it comes to CD remote controls is one that lacks index search buttons, and that's what we have here—absolutely unforgivable in a $2.5K player. Does Thorens imagine that high-end component buyers listen exclusively to rock/pop CDs? Another parsimonious touch is just one digital output, coaxial only, no optical.

My measurements revealed some serious shortcomings in this unit. To begin with, the frequency response starts rolling off at 4 kHz and is -0.4 dB at 20 kHz. That would be terrific in a phono cartridge but is a bit strange in a CD player. No big deal, but there's more. Full-scale THD + N hovers between -84 and -86 dB at most frequencies across the audio band, meaning excess distortion of 12 to 14 dB above the theoretical minimum of -98.08 dB. The quantization noise test yields -85.6 dB. Those are not good numbers. How much of the distortion is due to analog amplification became rather irrelevant when I looked at the monotonicity test pattern. There are D/A conversion problems here. The "stairway" goes up-down-up-down instead of up-up-up-up. That despite the typically perfect Bitstream gain linearity (less than 0.25 dB error all the way down to -100 dB), showing once again the limited significance of the gain-linearity test (see also David Rich on the subject in Issue No. 15, page 10). Wideband noise in the absence of a digital signal ("infinity zero") is quite high: -85 dB at 200 kHz and between -110 and -120 at most audio frequencies. Channel separation is pretty good: 86 dB at 20 kHz and improving to as much as 110 dB at the lower frequencies. On the other hand, a positive-going pulse on CD comes out negative at line out; such polarity inversion is extremely rare in today's CD players.

It is somehow my impression that Thorens wants you to plug the digital output of the TCD 2000 into the better-performing TDA 2000 processor (see below) and use the TCD 2000 only as a transport (i.e., CD "turntable"), with its straight-through mode relegated to a backup system. That would give you a $4300 CD player with two free digital inputs (one coaxial, one optical), one digital output (coaxial), two analog outputs (unbalanced only, one main, one backup), and no index search facility.

Any takers?

Outboard D/A Converter

Thorens TDA 2000

Thorens of America, Ltd., 84-05 Cuthbert Road, Kew Gardens, NY 11415. Voice: (718) 847-4289. Fax: (718) 849-7698. TDA 2000 outboard D/A converter, $1800.00. Tested sample on loan from manufacturer.

I am assuming that this unit represents Thorens's statement on the subject of correct D/A conversion. As such, it redeems the disappointing DAC of the TCD 2000 (see above) to a considerable degree. Indeed, if this were the only device available to me for playing my digital program sources, I'd be quite happy.

Unfortunately, the TDA 2000 has a minor defect, probably very easy to correct. Three LEDs on the front panel are labeled 32 kHz, 44.1 kHz, and 48 kHz to indicate the sampling rate of the digital input signal. Only the 44.1 kHz LED is ever lit, even when the input rate is 32 kHz or 48 kHz. At first I thought this defect was unique to my original sample, so I asked for a second one. Same thing. An entire production series appears to suffer from a manufacturing error. I found no other QC problems, however.

The TDA 2000 matches the size and styling of the TCD 2000; even the cockamamie wall-plug adapter is the same. The silicon complement appears to be Philips's best dual Bitstream push-pull chip set—I say "appears" because most of it is potted in two flashy Thorens-labeled modules and no circuit schematic was available. In this case full-scale THD + N measured a much more respectable -93 dB across most of the audio band, meaning 5 dB excess distortion above the irreducible 16-bit minimum. With the digital input level reduced, excess distortion of only 0.6 dB was achievable, which is close to perfection and proves the full-scale readings to be due to gain-related analog distortion. Increasing the word length to 18 bits and then 20 bits—since the system is claimed to have 21-bit resolution—resulted in a maximum improvement of 7 dB (just a little more than 1 bit) after excluding gain-related analog distortion. The 4 "marketing bits" don't bother me at all; I can live happily with a 17-bit processor.

Gain linearity is absolutely perfect with this particular Bitstream architecture; I measured ±0.0 dB error all the way down to -100 dB. I've never seen better. The "Rob Watts test" (FFT spectrum of a dithered 1 kHz tone at -60 dB) showed a bin-by-bin noise floor of -126 dB with a single second-harmonic blip of -105.8 dB. Not perfect but good enough. One little peculiarity: the frequency response has a peak of 0.1 dB at around 16 kHz; don't ask me why. My dogs can't hear it, and neither will you. I could have gone on trying to wring out this processor with endless other tests but I stopped. Why? Because (1) its basic performance characteristics were quite sufficient to establish it as a good, clean unit, and (2) I don't believe in outboard D/A converters anymore. Today's best CD players and other digital devices have good enough built-in DACs for the most demanding applications and have the advantage of being free of the possible pitfalls of the S/PDIF and AES/EBU interface. At this point the outboard D/A converter is being kept alive by high-end marketing rather than genuine need.

If you disagree with me, the Thorens TDA 2000 will most probably satisfy you and is recommended—as long as you need only a single unbalanced analog output because that's all you get. And, of course, they'll have to fix that sampling rate indicator.

❖ ❖ ❖

...and now, two little surprises from David Rich:

Budget-Priced Stereo Receiver

Pioneer SX-203

Pioneer Electronics (USA), Inc., 2265 East 220th Street, Long Beach, CA 90810. Voice: (213) 746-6337 [PIONEER]. Fax: (310) 952-2260. SX-203 stereo receiver, $225.00. No longer a current model; tested sample purchased by reviewer for $100.00 at Circuit City.

Yes folks, this is a review of a Pioneer receiver. "Pioneer receiver" is a dismissive term used by audiophiles to indicate lowly junk in comparison with their wonderful high-end stuff. Well, we know most high-end stuff is not wonderful but is instead the true junk. So, is the Pioneer receiver wonderful? Well, in many ways, yes. I paid $100.00 for the unit at Circuit City. I needed a receiver for an auxiliary system in my town house. For fun I also tried it in my main system, and it passed the ABX test, provided the level was held to below clipping. Before you declare me deaf, let's look inside to understand what is going on.

What is going on is that modern IC technology has allowed dramatic cost reductions to take place, just as in computers. The RIAA section is formed with an NJM4558 op-amp. It has much more gain and bandwidth than a low-cost discrete design found in old preamps. In better units you would see a lower-noise part, but I assume the designers are not expecting someone to connect an expensive moving-coil cartridge to this receiver. The tolerances of the passives that form the RIAA are not as tightly specified as in more expensive equipment, but modern manufacturing appears to keep them pretty tight anyway. From 1 kHz to 20 kHz the RIAA was flat within a 0.1 dB strip. It rose 0.5 dB from 1 kHz to 100 Hz and then backed off 0.1 dB when it hit 20 Hz.

Ergonomics of the receiver are very good. For example, you can directly enter a station frequency and tune to that station on the tuner. Speaking of the tuner, it uses a modern Sanyo chip (LA1836) for all functions except the FM IF and RF front end. With only two IF filters, this is a wide-mode-only tuner. If you have two close stations, forget it. The RF section is only a two-gang affair. The RF input is untuned. This is clearly intended for indoor antennas, with the hope that no local signal is going to overload the front end. Difficult signals are simply not going to be received, and many weak stations may come in only in mono. On the other hand, local signals can sound very good because that Sanyo chip has some state-of-the-art stuff. The VCO of the multiplex PLL is crystal-based (the $900 Pioneer Elite F-93 tuner we review in this issue does not have this feature). No birdie filter is required because of the Walsh-function-based multiplex decoder. On the downside, filtering on the audio path is minimal, so make sure you use the MPX filter on your tape deck.

One true pain about this product is that the 75Ω antenna input (the only antenna input) uses push-terminal connectors. I could find no adapter to connect to this. You have to strip the coax cable yourself to the bare ends and try to push these in the terminals. A sales person at Radio Shack (they sell rebranded versions of Pioneer receivers with the same problem) suggested using a 75Ω to 300Ω converter .That would have significantly decreased sensitivity and increased noise. (It looks like Radio Shack should change its motto to "You've got questions, we've got wrong answers.")

And now on to the line stage. The selector functions are done with Toshiba CMOS switches. Unlike switches made with older technology, these never wear out and are placed at the rear of the unit, so interconnect distances to the RCA jacks are short. Contrast this to the old mechanical selector-switch technology. Solid-state switching also adds to the good feel of this unit. Nothing about it says cheap.

The 300Ω resistors in series with each input prevent latch of the CMOS selector in case of a fault condition at the input. The tape monitors are not buffered, but 2.2kΩ resistors are placed in series with the record outputs as a low-cost solution to the problem of having the tape-outs loaded by the tape recorder when the recorder is off. Two record outputs are available. They are separately selected in the CMOS switch bank, so you cannot create self-oscillations like so many preamps we have reviewed that sell for ten times this receiver's price.

Another NJM4558 buffers the output of the CMOS selector switch. Then come the tone controls. You cannot defeat them, but the controls have center detents. These are well designed, including a treble control that uses a single transistor-based synthetic inductor. The active amplifier is another 4558. The volume control (motorized in the SX-203R remote version of this receiver) and balance control follow. These controls are buffered by yet another 4558, which also performs a bass-boost function for the kids that Pioneer calls Super Bass. Four electrolytics are in the signal path from line in to speaker out. Note that no gain is supplied by the line-level signal path. All gain comes from the power amp (41 dB). In the discontinued Pioneer SX-31, which listed at twice the price, Super Bass is out and a normal loudness function is in. The stage that does the loudness function (which is defeatable, as are the tone and balance controls) has 20 dB of gain. The power-amp gain is reduced to 20 dB in the SX-31.

The SX-203 power amplifier section is all discrete and has a topology that for the most part is like that of a megabuck power amp. A differential pair with resistor tail and resistor load drives a current-source-biased second stage with a V_{BE} multiplier for the bias circuitry. A complementary pair of emitter-follower predrivers then drives the single complementary output pair. A higher-end circuit design in some slightly more expensive receivers would include a current mirror load for the first stage (as done for example in the Pioneer SX-31 referred to above) or a complementary drive to the second stage, using a unity-gain inverting stage. Other circuit tricks, such as a bias circuit that prevents the nondriving device from fully turning off, might also be included. Again, we are not talking megabucks here because the Pioneer SX-31 adds just such a circuit. One problem with the power amp in the SX-203 is the small size of the capacitor in the feedback loop (22 μF). This results in a pole at 8 Hz and as a result the frequency response is down 1 dB at 20 Hz. The -0.2 dB points are 70 Hz and 22 kHz. Between 200 Hz and 10 kHz we are talking ±0.025 dB. The top-end rolloff occurs because of the high gain levels in the power amp. If the gain had been less, the bandwidth would have been greater.

The big difference between the SX-203 and the bigger well-designed power amps is the iron. The transformer puts out about 55 V but it has a high winding resistance that limits current capacity. Heat sinks are small, as is the full-wave rectifier. The single pair of output devices is not designed for operation at high current levels, and 8200 μF capacitors are all that are used for the primary filter capacitors. Power for the low-level circuits comes from a zener-diode regulated supply for the -12 V rail (are they following high-end practice?). The +12 V rail gets an IC regulator. Separate half-wave rectifiers on separate supply taps of the transformer supply the unregulated voltage for the low-level circuits.

The layout of the receiver is well done because crosstalk stays below 60 dB for frequencies lower than 1 kHz. At 20 kHz the worst-case number is 37 dB from line in to speaker out at a 1-watt level. I can name many high-end products that do much worse.

The SX-203 has a complex dc and overdrive protection circuit with 6 active components, 27 passives, and a relay at each speaker output connection. No el cheapo single-transistor foldback circuit here. Such circuits are found in most high-end products, but Pioneer left them behind long ago. As a result, the PowerCube looks very good. The voltage supplied into reactive loads increases relative to the resistive load, just as you would expect of a well-designed amp. The unit had no problem driving 1Ω loads on a dynamic basis. Can your single-ended triode do that, tweaks?

Into an 8Ω resistive load the amplifier of the SX-203 will deliver 31V rms. That declines to 25 V rms into 4Ω, 17 V into 2Ω, and 9 V into 1Ω. These numbers are limited by the current this amplifier can supply. The limit is 9 amps. As I said, it is the iron that got thrown overboard in this design. You may not be able to drive a high-end speaker with a crazy impedance, but most speakers will be driven to very loud levels by the SX-203. One must note that this unit will not pass 4-ohm FTC power tests because it cannot pass the preconditioning tests, again as a result of the missing iron and output transistors. Lack of an FTC 4-ohm rating on any piece of audio equipment is a sure indication that it is underdesigned. Meeting this test requires that some real money be used in the design.

That skimpy power supply shows up in the distortion measurements. With one channel driven, the amplifier drops to a THD of -70 dB. It clips just above 110 watts at 1 kHz and 20 kHz. No dynamic distortion is seen in the 20 kHz measurement (do you see red faces on some high-end amplifier designers?), although it may be hidden by the static distortion level. The 20 Hz distortion starts rising at the 40-watt level—half the value of the 1 kHz and 20 kHz conditions. Here the power supply limitations are starting to show. As the frequency gets lower, more current must be supplied at the maxima and minima of the sine waves because these peaks and dips exist for a longer period of time. The filter capacitors get drained and the power supply collapses. With both channels driven, the power levels at 1% (-40 dB) distortion drop to 90 watts at 40 Hz. At 20 Hz only 80 watts are produced. Overall, distortion becomes 5 to 10 dB worse with both channels driven. Dropping the load down to 4Ω with both channels driven produces a 20 Hz to 20 kHz power level of 100 watts at 1% distortion, and a best-case distortion number of -60 dB (0.1%) at 70 watts. Pioneer rates this as a 70-watt-per-channel unit at 1% distortion into 8Ω. It clearly exceeds this specification. Can all the high-end amplifiers claim to meet their specs?

The SX-203 may represent a little too much slumming for most audiophiles because of its limited power supply and output stage, as well as the bandlimiting in the power amplifier because of its high gain. Most of these problems appear to be resolved in the somewhat more expensive SX-31. But the SX-203 does some remarkable things on its own, including driving complex loads down to 1Ω, showing no dynamic distortion, and producing good crosstalk numbers. That is why it has no perceptible sonic signature within its power limits. In many respects this receiver puts much of the high-end equipment to shame. Professional engineers designing to specifications win over tweaks designing to an agenda. We intend to spend more time in the bargain basement in future issues to see how much one really has to spend on electronics.

This unit is discontinued now, but I do not think it is a magic design; in fact, it is fairly typical. Most major-brand Japanese receivers are of basically similar design. Look for a discrete power amp. You also want a direct-path switch to bypass the cheap balance and tone pots. FTC power ratings down to 20 Hz and THD numbers below 0.1% are also good signs. *Consumers Reports* can also help steer you to a good unit, especially when it comes to tuner performance. They test more samples of low-end receivers at one time than I can ever attempt to.

* * *

Editor's Note: *The SX-203 was dropped from the Pioneer line shortly after Issue No. 23 came out, but we figured it was still worth testing for this issue as an illustration of what is obtainable these days for peanuts. The SX-205, at $220.00, appears to be an improved replacement. As for ABX double-blind listening comparisons, I set one up in our lab against a $3K power amp canonized by* Stereophile *(in a slightly different version) as "Class A." Three highly experienced audiophiles, of whom I was one, compared the two amps for hours. The correct blind identifications were the usual 50%, give or take a couple of points—in other words, completely random. Interestingly, my two fellow auditioners (not I) actually expressed a preference for the SX-203 as long as it was known as B. When they switched to X, however, they got nowhere. To me the outcome was predictable, but the test had to be performed because until then we had never ABX-ed really cheap electronics against the multikilobuck high-end stuff. Sound taps for the tweako belief system...*

Integrated Amplifier
Yamaha AX-570

Yamaha Electronics Corporation, USA, 6660 Orangethorpe Avenue, Buena Park, CA 90620. Voice: (714) 522-9105. Fax: (714) 670-0108. AX-570 integrated stereo amplifier, $499.00. Tested sample on loan from manufacturer.

Consider my Pioneer SX-203 review above. While it is remarkable for what it is, the Pioneer has limited current-sinking capability, sloppy RIAA equalization, a low-frequency rolloff in the power amp, etc. It does not have a tone defeat switch or a record selector switch, and you cannot separate the preamp from the power amp. The logical question is—what can you get if you take one step up from the bottom of the line? Well, in receivers the next step up turns out to be a multichannel receiver of about the same quality and left/right-channel performance. If you then look into integrated amplifiers, you may very well be surprised to find out that there is no inexpensive integrated amplifier in the line. (Yes, I know, the situation is different in Europe.) Which brings us to the Yamaha AX-570. At $499 it represents about the least you could spend without making significant compromises.

Build quality is good enough to ensure long years of service but it is not in the audiophile jewelry class. The unit is made in Malaysia. The chassis uses thin metal held together with sheet-metal screws; resistors are carbon; the switches are not sealed; the PC boards are single-sided and filled with jumpers; indeed, on some parts of the board large 10-gauge jumpers run above 20-gauge jumpers mounted directly on the PC board. (It looks like some sort of multidecked highway structure.)

The audiophile—actually, industrial—class of design really does cost a lot more, and forcing the prices even higher is the fact that those high-end units sell at low volumes, since most people do not see the need to spend that kind of money in order to own equipment that will last 20 years and look as if it were ready to fly on the space shuttle. A car analogy may help at this point. The Pioneer SX-203 is a Ford Escort. The Aragon, Bryston, and Mcintosh are the BMWs of the audio world. The Audio Researches and Conrad-Johnsons are equivalent to suppliers of reconstructed Ford Model T's. The Yamaha AX-570 is the Toyota Camry of the audiophile world.

Given the thin sheet metal, most of this unit's 24 pounds come from the transformer and the heat sinks (separate for each channel). Mounted on each channel's heat sink are four output devices. Each side of the class A/B output stage uses paralleled devices. This unit does have the ability produce some current, unlike the Pioneer, which has a smaller transformer and heat sinks and does not have paralleled output stages. Of course, even more current is available in separate power amps that have even more iron. In most cases the voltage and current capabilities of the AX-570 should be more than adequate.

The rest of the power amp shows similar thinking. The all-bipolar circuit uses a differential pair with active loads. An emitter follower connects the signal from the input stage to the single-ended second voltage-gain stage with a single-transistor current source. Another emitter follower follows this stage and it drives the aforesaid pair of output devices. The only electrolytic capacitor in the power amp is at the input. This design may not produce the lowest distortion theoretically possible, but the distortion levels are surprisingly low, only marginally higher than the best we have measured on any unit and in any event well below the distortion levels produced by some megabuck amplifiers canonized by the high-end crowd.

The power supply for the power amp has 12,000 μF filter capacitors. The output devices are monitored with an overcurrent protection circuit similar to that of the Pioneer SX-203. When overcurrent conditions occur long enough to cause the output transistors' safe area of operation to be exceeded, the relays in series with the output are opened and the signal at the input of the power amp is muted. Unfortunately, our PowerCube test system was temporarily down when the AX-570 was on the lab bench, so we were unable to obtain a complete picture of maximum output into the widest range of resistive and reactive loads, but we did determine that it took a little over 11 amperes of steady-state current at 1 kHz to make the amplifier go into protection. That's not half bad.

There are a separate relays for speaker pairs A and B (which have full-size banana output jacks, not the cheap connectors used by Pioneer). This eliminates the need to route the output of the power amp back to the front of the unit where the speaker selector switches are. The switches just drive the relays. One interesting idea unique to the AX-570 is that the ground at the speaker terminal is sensed and compared with the ground used at the low-level electronics. The output of the op-amp forms the ground reference for the power amplifier. This approach ensures that no hum or distortion is introduced by drops in the ground line that occur when signal current flows through ground wires connecting the speakers to the power supply. It also functions as a dc servo, eliminating the electrolytic cap in the feedback path. Another nice touch along a similar line is the placement of the preamp's gain stage on the same auxiliary PC board that supports the volume control. This allows a low-impedance signal to be routed to the power amp.

The op-amp used is actually a pair of op-amps. Both sections of an NJM2068S are used for each channel. Each op-amp is wired with a pair of resistors as a noninverting amplifier. A pair of 470Ω resistors sums the signals from each op-amp. This approach improves the noise of the circuit by 3 dB. Using a lower-noise op-amp would appear to be a simpler approach, but it looks like the economies of scale involved in mass purchases of the NJM2068S makes acquisition of a separate op-amp for

(continued on page 64)

FM Tuners: The Continuing Survey

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

Not all tuners are created equal, not even in the same price range. Here is the opportunity for the audiophile to find genuine performance differences. Or are amplifiers that much more interesting?

This is the second part of our tuner survey. We intend to have a third part to bring in a couple more models but we pretty much have all the top-of-the-line production tuners covered between these two installments. That tells you how high-priced tuners are selling these days.

Your Editor has asked me to give some indication of how a high-end tuner should be designed. The question implies that the designers of these tuners know less than I do, which is not the case. Each of the tuners has been designed to emphasize the specifications felt important by the designer, under the constraints of how large the bill of materials would be allowed to get. The ultimate tuner would cost somewhere around $3000 to $5000 these days. Day Sequerra is working on such a design for introduction in the summer of 1997, but no others are on the horizon because manufacturers believe they will never recover the cost to create the tuner, given the falloff of this market.

James Bongiorno has shown me his prototype design and it looks really really good on paper, but as of this writing it is a piece of paper and may remain so. (Bongiorno also has some novel amp and preamp designs that stand a much better chance of seeing the light of day. He also showed me some of his work that predates my involvement with *The Audio Critic.* I would have praised these designs for their innovation if I had known about them.)

Having hedged sufficiently, I'll tell you what I would do if I were assigned to design a super tuner. My priority would be to design a tuner that minimized adjustments. Almost every tuner we have seen has been out of alignment. I would implement everything past the FM demodulator in a digital signal processor. Keeping the demodulator in analog form makes the requirement on the A/D similar to audio A/D. I should note that work on bringing the demodulator into the digital domain is very much a current research topic, with some promising results, but for this discussion let's keep things simple and digitize after the demodulator.

The passband of a composite FM signal runs out to about 60 kHz, so the sampling rate needs to be higher than for an audio ADC, but the dynamic range is a lot less so we can still use the same front end of the delta-sigma modulator but we need a different decimator. Once we have our signal decimated, we can perform all sorts of mathematical operations that are very difficult in analog land. The neat ideas of the Pioneer Elite F-93 (see below) could be made to work reliably in the digital domain, for example. Properly done, signal-to-noise ratios, distortion levels, and channel separation figures would be much better using the DSP approach. Once the DSP has produced the stereo signals from the composite signal, the data could be sent out of the tuner as an S/PDIF signal or it could be converted to analog by an internal pair of DACs.

So what about all those adjustments in the front end, IF strip, and demodulator? (The latter has adjustments only if we go for the ultimate performance of the PLL demodulator instead of the zero-adjustment pulse-count detector.) Well, once we have DSP on board we can do self-test and alignment. The DSP engine can do level measurements across frequency and it can do distortion and noise measurements. The DSP can be connected to DACs that can generate dc voltages to drive varactors to change the resonant frequency of tuned circuits. The DSP in conjunction with a DAC and an up-conversion mixer would form an RF test generator that could be used for the autoalignment and self-test process. Research papers have been published on the use of these techniques in cellular phones.

OK, time to stop dreaming and look at the reality of what is available today.

(Please note that all stations discussed in the use tests were tuned in and received at Richard Modafferi's test facility near Binghamton, New York, unless otherwise noted. The number of these use tests has been increased this time around. Additional details are in the reviews.)

Accuphase T-109

Accuphase Laboratory, Inc., Yokohama, Japan, through Axiss Distribution, Inc., 17800 South Main Street, Suite 109, Gardena, CA 90248. Voice: (310) 329-0187. Fax: (310) 329-0189. Model T-109 quartz-lock synthesizer FM tuner, $2995.00. Tested sample on loan from distributor.

(The following is a fly-on-the-wall view of the Accuphase tuner test at Richard Modafferi's lab.)

Hey, Rich, for three thousand dollars this had better be the world's best tuner. Oh, you tested one already and you think it *is* the world's best. I guess that's proof that there is no free lunch in this world. Yes, Rich, I tried to get schematics but Accuphase would not send them, so let's get directly to the use tests. What do you mean I'm going to have to wait for you to realign the front end? How could a three-thousand-dollar tuner be out of alignment? Wait a minute, you are saying you also have to touch up the stereo separation trim pots? Well, at least you won't have to do anything to the FM detector because pulse-count demodulators require no adjustment.

Finally, you're finished. Let's try that 3/4-wavelength indoor antenna first. Gee, it gets signals 120 miles away. OK, let's connect it up to the antenna tower. Hey, we can get 91.3 MHz with a little splatter even though it is at 100 µV and 91.5 MHz is at 30 mV. Yes, Rich, I know your dear old MR-78 gets this signal cleanly but Mcintosh doesn't make that one anymore. Look, we have a very slight cross-modulation caused by the 1 V signal on 92.1 MHz. Remember, the Onkyo and the MR-78 are clean in the local mode but they have some cross-modulation in the DX mode. And note the Accuphase has a $2f_1 \pm f_2$ spurious at 106.3 MHz (f_1 and f_2 are 105.7 and 105.1). Despite this, you can hear a weak signal at 106.3 MHz on the Accuphase, but no signal is heard on the MR-78 and Onkyo.. Let's try the killer 89.5 MHz test (140 miles away). That's the station surrounded by 89.3 MHz (4 miles away) and 89.7 MHz (2 miles away). Looks like the Accuphase has some splatter. Yes, your MR-78 is better than the Accuphase on this test but not by much.

Hey, Rich, I told you we don't have the schematic but you can reverse-engineer it if you want to. What do you mean it is a very ordinary design? So you found double-tuned RF circuits are used on the RF input and mixer input. What else did you find out? The RF transistor is a JFET 2SK241 and the mixer and oscillator is a 2SC2668. But that's just ordinary semiconductors. Rich, look again, it says right here in the literature on this thing that it has a double-balanced mixer, so you missed a transistor. You're sure it has no double-balanced mixer? But Rich, this front end has excellent noise quieting, very good spurious rejection, and therefore wide dynamic range, so something must be different. Oh, it is just good design with ordinary parts. So that's why RF folks get paid so much because RF design is a black art. While you are in there, what multiplex decoder chip does it use. Sanyo LA3401? Oh, that's the one that Mcintosh uses. I wonder why they did not use the better LA3450.

OK, let's put it on the bench and see what it will do. Look, the 1 kHz stereo THD is only -74 dB in the wide mode and your stereo IM test is also at -74 dB. Aren't those the best you ever measured? Separation looks good too, with 50 dB at 1kHz and 36 dB at 10 kHz. And look, the broadband noise is below -80 dB. The narrow mode is not as good, but this is a very narrow filter similar to the one in the Onkyo T-9090II. Oh well, 22 dB channel separation is OK across the band, as is a 1 kHz THD of -46 dB and a stereo IM of -49 dB, but doesn't the Pioneer Elite F-93 do better than that? (OK, I know, the F-93 has a bad RF section, so it doesn't matter how well it does in the narrow mode, and its wide-mode performance was real bad.) Rich, did you know that Accuphase guarantees its specifications? Some of them are a little better than what you measured—maybe you need a better FM generator. OK, please calm down; I will not say anything else about your FM generator, and I know the sample we got did not come anywhere close to specs until you aligned it.

Look, this multipath meter function is really useful. It measures the AM modulation on the signals that are caused by multipath and interfering adjacent signals. I bet my cable FM will make the needle move out of the good zone. Rich, you have any idea why they could not give you a separate meter for signal strength instead of using this pushbutton switch to go between the two functions? It does cost $2995, you know. Yes, I know, it has a separate meter-circuit IF, so strong signals don't just pin the meter, but so do the Onkyo and the Elite, and they cost a lot less. I know, I know, they don't have the look and feel of this unit. It does look a lot more expensive, but I don't see three thousand dollars' worth of parts in there, even if it has higher build quality than Onkyo and Pioneer. Why is it built with single-sided PC boards? Yes, I know, the metalwork on this thing is very expensive, and that big shielded power supply is not something you see every day.

By the way, did you notice that if you tune in a weak signal in the wide IF mode, the stereo/mono indicator goes out? You cannot tell if it's stereo or mono. Looks like a logic error to me. Yes, yes, the problem does not happen in the narrow mode, but this thing is three thousand dollars. And did you notice it has no antenna selector? Oh, and do you know why it beeps every time you move the tuning knob that's connected to the rotary encoder? You say it's for blind people. Very interesting, but I wish they'd give you a defeat switch. I don't like to be beeped at. Did you notice that the tuning knob has such a great feel because they put that big flywheel on it? And at least they give you a remote control and balanced outputs for your $2995.

I wonder how long we can keep it. What's that?

Peter is on the phone and he wants the tuner sent to the home office as quickly as possible? You didn't tell him it was any good, did you? Yes, I know, it *is* the world's best production tuner, and you know it's the world's best, but Peter didn't have to know!

Followup (Discontinued Model)

Harman Kardon TU9600

Harman Kardon Incorporated, a Harman International Company, 80 Crossways Park West, Woodbury, NY 11797. Voice: (516) 496-3400. Fax: (516) 496-4868. TU9600 "active tracking" AM/FM stereo tuner with remote control, $449.00. Tested sample on loan from manufacturer.

This is the followup report on the TU9600 I described and analyzed (qualitatively) in the last issue. Since it is now discontinued, with no tuner of comparable quality in the current Harman Kardon line, you will find out whether you got a bargain if you bought one or missed a bargain if you did not.

The RF section is no bargain, with single-tuned circuits at the input of the RF and IF. This is typical at this price point. An earlier, more expensive, and also discontinued tuner (Citation 23) had double-tuned circuits. It came as no surprise that the external-antenna torture test resulted in the entire dial being filled with spuriae from the 1 V signal at 92.1 MHz and the 250 mV signal at 105.7 MHz. Weaker 100 mV signals were also audible as spuriae. But do not get too depressed; with an indoor or simple outdoor antenna you are not going to get the same signal levels as Richard Modafferi does. What you want is sensitivity and the TU9600 has that in spades. Richard got a mono signal form Canada that was 220 miles away. At my place in Pennsylvania the TU9600 did as well as any other tuner I have had my hands on, as long as it was in the Active Tracking mode. Using the plebeian quadrature detector was a disaster, however. One very minor problem Richard found in the Active Tracking mode was that a very strong AM signal (measuring 5 V/m, from a station 600 feet from his house) could be heard at a dead spot in the FM band. If an FM signal were present, it would override this spurious.

One neat feature of the Active Tracking circuit is the ability to create a static offset at the VCO input. This shifts the PLL lock range. A close interfering signal may thus be ignored by the PLL when the lock range is prevented from extending into the band that the interferer occupies. This feature is called Fine Tuning by Harman Kardon. We did not find this feature helped under our signal conditions but it might help under yours. It is a very clever, low-cost idea.

The IF strip has three ceramic filters. The IF amps are one-transistor discrete circuits. Such a circuit may be less expensive than an integrated-circuit IF amp. No phase adjustments are available in the IF strip. The result of this (and perhaps misadjustments in the quadrature detector) is higher distortion, -54 dB in the normal mode and -48 dB in the Active Tracking mode. Do not feel too bad, though; the $900 Pioneer Elite with the world's most complex detector circuit has only 2 dB better distortion. The Sanyo LA3450 multiplex decoder did strut it stuff. Stereo IM was a state-of-the-art -75 dB in normal mode, dropping to a very good -60 dB in Active Tracking. Channel separation was better than 40 dB across the board. The frequency response of the tuner was up about 1 dB at 15 kHz. AM was typical for tuners of today, which is to say an ancient five-tube table radio would blow it away.

This little tuner even includes A/B antenna switching, as well as high blend. As I said, in the Active Tracking mode it did as well as anything in my system. If it were still around, I would include it under my recommendations. I hope Harman Kardon, makers of a number of great tuners in the past, will replace this unit as well as the more expensive Citation 23.

Followup

Mcintosh MR7084

Mcintosh Laboratory, Inc., 2 Chambers Street, Binghamton, NY 13903-2699. Voice: (607) 723-3512. Fax: (607) 724-0549. MR7084 AM/FM stereo tuner, $1500.00. Tested sample on loan from manufacturer.

Mcintosh's only currently made tuner was introduced and given the once-over by the Editor in the last issue. This is the promised followup after use tests and measurements.

The FM has an audible 500 Hz beat note on signals from 87.9 to 90.0 MHz. It is about 50 to 60 dB down. It may be a PLL loop-filter problem in the electronic tuning system. FM selectivity was good enough to get a station 140 miles away at 91.5 MHz between a station 75 miles away at 91.3 MHz and a station 50 miles away at 91.7 MHz. This test was done at night, when the station that is 4 miles away at 91.5 MHz is off the air. We run the test with that station *on* the air when testing the super tuners. With the MR7084, forget it; its selectivity is not good enough. The tuner could get a station 220 miles away, showing it has good sensitivity if not selectivity. No spurious problems occurred with a 250 mV signal at 105.7 MHz. A weak signal at 105.3 MHz could not be tuned in because of the lack of selectivity (recall the tuner has no narrow IF mode) but not because of a spurious problem that affects so many other tuners here. With a 1 V signal at 92.1 MHz cross-modulation at 91.7 MHz is audible, but it looks like that may be a problem with the tuner's sensitivity, not the RF section. We look for a $2f_1 \pm f_2$ spurious signal (f_1 and f_2 are 105.7 and 105.1 MHz) that can cover a weak station at 106.3 MHz. The weak spurious signal at 106.3 MHz could not be heard but 106.1 MHz came in noisily at that location. Insufficient selectivity prevented us from seeing if the spurious signal was present. The 106.1 MHz station was clean when we tuned to it because of the good spurious response rejection of this tuner. Other tuners might not get this station. Given the ordinary components in the RF section, performance is remarkably good. Only the Onkyo T9090II, Accuphase T-109, Yamaha TX-950, and Rotel RHT-10 are in the same class. This again shows that superior RF design is achieved by good engineering and black magic. Overall, our use test of the MR7084 showed that the RF spirit is willing but the IF flesh is weak.

On the bench, the THD performance was good, with 1 kHz stereo distortion at -63 dB and 10 kHz IM at -67 dB. What's more, channel separation was state-of-the-art, measuring better than 50 dB across the audio band. Maybe those IF filters do have superior phase characteristics, just as Mcintosh claims.

The AM section was badly misaligned. Once aligned, AM performance was very good. A 1560 kHz station 220 miles away came in with no spuriae, even with a 1430 kHz station broadcasting just 600 feet from the test site.

In the area of ergonomics, the preset pushbuttons have a 1 to 2-second delay, which is a pain. Features such as A/B antenna switching, rotary knob tuning, and a serious signal-strength meter are absent here. It is hard to recommend a tuner that has a low whistle on stations below 90 MHz and lacks a narrowband IF mode. If you listen to well-spaced stations above 90 MHz and have always wanted a Mcintosh tuner, this might be the one for you, but I would consider a used MR-78 instead. In most cases it will cost less and is so much better than the MR7084.

Pioneer Elite F-93

Pioneer Electronics (USA), Inc., 2265 East 220th Street, Long Beach, CA 90810. Voice: (213) 746-6337 [PIONEER]. Fax: (310) 952-2260. Elite F-93 AM/FM stereo tuner, $900.00. Tested sample owned by reviewer.

Pioneer has an interesting philosophy in the design of top-of-the-line tuners. With each generation they make a major advance—but they remove the advance they made in the last tuner. Take the Pioneer Elite F-99X (and the similar Pioneer F-90) introduced in 1985. It had a highly innovative baseband signal processor using two Pioneer ICs (PA5006 and PA5007). A pulse-count demodulator controlled a switch that chopped a 38 kHz sine-wave carrier signal. This system was discussed in Issue No. 23. *Audio* magazine (November 1985) measured this tuner's stereo THD to be -76 dB at 1 kHz and -60 dB at 10 kHz. Stereo separation was 53 dB at 10 kHz. In 1988, the F-91 was introduced. This had the active-tracking IF strip that offered sharp transition bands and good phase characteristics (again, see Issue No. 23). In addition, it allowed the station to be slightly detuned without incurring a distortion penalty. But for some unknown reason Pioneer had removed the stereo decoder they had had in the F-99X. Instead, they used a PLL demodulator and what appeared to be an analog multiplier for generating the L-R signal. By doing this they retained the approach they had used in the F-99X, demodulating R+L and R-L, and then forming the sum and difference to get L and R. As explained in Issue No. 23, a switching MPX decoder gets L and R directly from the composite FM signal. The sinusoidal 38 kHz carrier tone required when an analog mixer is used to generate L-R came from the same chip as used in the F-99X (the PA5006). The result of the change in MPX decoder technology was THD which measured twice to five times that of the F-99X, as reported in the August 1988 issue of *Audio.* Strangely, no mention of this backward progress was made in the review.

Now we have the F-93 with a novel stereo noise-reduction filter and stereo decoder, but the active tracking IF is gone. Also gone is the PLL, replaced by a (oh no!) quadrature detector. Actually, three quadrature detectors. Two Pioneer PA5008 FM demodulator chips are run parallel to reduce noise by 3 dB. (The Sanyo LA1235 appears to be a lower-noise device, and one LA 1235 does the job about as well as any quadrature detector can be expected to.) The analog multiplier block for the generation of the L-R signal is also in the Pioneer chip (PA5008), as are the final stage of the IF strip, the mixer for the quadrature detector (for L+R generation), the signal meter circuit, and the mute circuit (these last items are also part of the LA 1235 and similar chips).

In the PA5008 the analog multiplier actually has three inputs: one for the output of the IF strip, one for the delayed signal from the LC filter that is part of the quadrature detector, and one for the 38 kHz sine-wave tone. This suggests that this block demodulates the L-R signal directly from the IF signal by combining the mixer function of the quadrature detector with the analog multiplier function required to decode L-R (I think this is what Pioneer means by direct decode), but I do not have the circuit details of the PA5008 to say exactly what is going on inside the chip. The other mixer in the PA5008 produces the L+R signals from the output of the IF strip and the delayed signal from the LC filter. This is the standard approach for a quadrature detector.

The third quadrature detector employs a strange distortion-canceling system. This third quadrature detector uses a much simpler phase-shift circuit that results in more distortion. The output of this quadrature detector is subtracted from the main quadrature detector output (coming from the two paralleled PA5008 chips) to yield the distortion. This distortion is then summed into the left- and right-channel audio signals at the output of the tuner (after L+R and L-R have been combined to make L and R). A pot adjusts the level and polarity of the distortion being introduced. The object appears to be to cancel some of the distortion in the main signal path. Separate pots are included for narrow and wide mode. But wait, there is more kludge to come! Another set of distortion-reduction pots forms a feedback loop from the output of the analog multiplier in the PA5008 back into the input port along with the 38 kHz carrier signal that is normally connected to this port. The pots are adjusted for best distortion. The exact function of the feedback is unclear, since I do not have a schematic of the analog multiplier inside the PA5008 chip. Different pots are used for the narrow and wide mode.

You would think this complex mess would not work over time and temperature, and you would be correct. *High Performance Review* (Winter 1991-92) measured between -60 dB and -54 dB stereo THD out to 3 kHz. At 10 kHz the number was -40 dB (1%). Once again, a two to five times doubling of the THD over the F-91, or four to ten times over the original F-99X. Now, I thought it was possible that the *High Performance Review* unit had been out of alignment, so I asked Pioneer to align our test sample before shipping it out. It did not help. Our sample measured -56 dB at 1kHz in stereo and our 10 kHz stereo IM test was at -54 dB. These were just about the poorest numbers we got in this survey. Stereo separation was 49 dB at 1kHz and only 36 dB at 10 kHz. That is better than *High Performance Review's* 32 dB and 28 dB, respectively, so it looks like our sample was better aligned, but even the aligned numbers are much worse than those obtained with the old F-99X.

Let's summarize. In 1985 Pioneer produced a tuner with state-of-the-art FM demodulator and MPX demodulator performance, even by today's standards (only the Accuphase matches it—see above). It had 5 adjustments. The current product has performance that would have been average in 1980. It requires 14 adjustments and uses twice the number of components. Is something wrong with this picture? Pioneer must have thought so because they use a Sanyo chip in their new receivers (see my SX-203 review elsewhere in this issue) instead of their own parts. That Sanyo chip has a higher-speed, adjustment-free VCO that is missing on the F-93 MPX decoder chip (the original PA5006 from the F-99X). It is interesting to see that this low-end Sanyo chip can take on the complex F-93 setup. For example, I had a station with a low-level 19 kHz pilot tone that would not go into stereo on the F-93 but did so with the Sanyo-based tuners we were testing.

One nice thing on the F-93 is the use of individual pots to adjust separation in wide and narrow mode. Another nice thing is that the MPX filters in the audio chain are a GlC-based 5th-order elliptical ladder design using 4 op-amps. That kills the 19 kHz pilot and subcarrier very well. *Audio* reported problems with the F-91, which had a simpler 2nd-order circuit with one op-amp. *Audio*, however, in the typical fashion of the commercial press, reported that they suspected the problem "was peculiar to [their] sample." The circuit used in the F-91 provided only a theoretical 6 dB of rejection at 20 kHz. Even if the pilot-tone canceler was misadjusted, it would have taken divine intervention to make any sample of the F-91 have good subcarrier product rejection.

The RF front-end design of the F-93 looks more promising than the demodulator and decoder, with double-tuned filters for both the RF stage and the mixer. The mixer is double-balanced, with two TV-tuner front-end chips used in the mixer section. Unfortunately, the power supply voltage for the RF stage is 8.5 V, which is lower than in most tuners (usually 12 to 15 V). Perhaps that was why the tuner's performance on our outdoor antenna torture tests was so poor. Another reason may have been that the F-93 does not modify the RF circuits in the local mode as Rotel and Onkyo do but instead has just a 4-position signal attenuator at its input.

Our 1 V signal received at 92.1 MHz caused serious cross-modulation at ±0.5 MHz from the signal frequency. A signal at 91.7 MHz could not be heard clearly. The 91.3 MHz signal that is only 100 (µV and adjacent-channel to the 30 mV local signal at 91.5 MHz was receivable but of only fair quality. The 250 mV local signal at 105.7 MHz caused cross-modulation from 105.1 MHz to 106.9 MHz. Activating the RF attenuator eliminates this, but a weak desired signal will also be attenuated. With the attenuator on, we were still able to receive a 10 µV signal at 105.3 MHz. A serious $2f_1 \pm f_2$ spurious occurred at 106.3 MHz (105.7 MHz interacting with 105.1 MHz, which could not be received). We did not find $2f_1 \pm f_2$ spuriae resulting from the even stronger 92.1 MHz, however.

The front-end problems made it impossible to see how well the very complex IF filter worked because the worst-case desired signals were covered with cross-modulation and spurious signals. The IF strip has 8 (count them, eight) filters and amplifiers in the signal path. The first 3 filters are swapped in narrowband mode. I strongly suspect that the last 5 filters are used as band-pass limiters, accounting for this tuner's very high AM rejection of 80 dB. (This is the manufacturer's spec. We did not measure AM rejection.) That specification just would not happen if the tuner had a simpler IF and quadrature detector. Somebody at Pioneer is clearly familiar with the work of Baghdady, since the concepts of narrowband limiters and tracking filters in FM both originate with him. I still cannot understand why Pioneer punted on the tracking bandpass-filter-based IF in the F-91. They claim part of the problem was the number of adjustments required to make it work, but the F-93 IF strip also requires lots of adjustments. It has 6 phase-tweaking adjustments, 3 for wide and 3 for narrow, to correct the phase response of the strip. The F-91 did not need to be phase-tweaked.

On our indoor antenna tests in Pennsylvania the tuner did as well as the other super tuners in this survey because the smaller signals did not overload the front end. The IF strip in the narrow mode brought in weak adjacent-channel signals as well as, but not any better than, the rest of the pack.

OK, is anything really good about this tuner? Well, yes, it has a proprietary MPX noise reduction system that is the only serious attempt, other than Carver's, to deal with noisy FM signals. In the Pioneer, as explained in the last issue, the L-R signal is split into 8 bands. A voltage-controlled amplifier (VGA) block determines how much L-R signal in each band will be subtracted from the L+R signal when the final L and R signals are formed. The VGA is controlled by a noise detector. The more noise in the band, the less L-R signal is allowed out. Two LSI ICs and 18 op-amps are needed in this circuit. Those LSI circuits each have another 22 op-amps plus the 8 VGAs. The good news is that it works. Noise is reduced much more than with a high-blend circuit, and the stereo effect appears to remain intact. The bad news is that even though noise is dramatically reduced, the signal is still more distorted than it is in mono, and you can hear the signal clean up if you switch to mono. How important this feature is to you depends on how important stereo on weak signals is to you. This is the only super tuner that has this kind of function now that the Carver TX-11 is discontinued.

Another neat trick on the F-93 was also explained in Issue No. 23. The L-R signal can be generated using only the lower sideband of the subcarrier in the F-93. This is done using a single-sideband demodulator. Pioneer calls this the S-MPX mode. In a normal double-sideband demodulator you multiply the carrier (for FM stereo the 38 kHz sine wave generated from the 19 kHz pilot tone) with the signal to be demodulated (the composite FM signal). In a single-sideband demodulator you do that and then add in a 90° phase-shifted version from a second multiplier. The input to this multiplier is the composite signal and a 90° phase-shifted version of the carrier. It is very difficult to phase-shift the broadband multiplier output (it is easy for the single-frequency 38 kHz signal), and a circuit that requires 12 op-amps and 38 passives is used in the F-93. The two PA5008 chips are used for the two multipliers. Instead of being paralleled together as they are in normal operation, one PA5008 gets the inphase 38 kHz signal and the other gets the phase-shifted version. A simple one-op-amp circuit generates the 90° phase shift of the 38 kHz signal in the S-MPX mode. In the normal mode the circuit is reconfigured so that no phase shift occurs, allowing the second PA5008 to be paralleled with the first (when in the S-MPX mode, the L+R of this second PA5008 goes unused). True nerds who get the service manual will note errors in the schematic around the 38 kHz phase shifter. You have to trace the board to figure out how it works.

Now we get a clue to what this tuner is about. It looks like the S-MPX circuit was designed first and then they backfilled to get the normal circuit. The normal circuit does not work so well, but the S-MPX is excellent when considering its job of bringing in weak signals with nearby interference. With the S-MPX and narrow filter enabled, we obtained 1 kHz THD of -44 dB, and the 10 kHz IM was -50 dB. Stereo separation was also very good, running about 44 dB at 1 kHz and 36 dB at 10 kHz. These are very impressive results considering that the upper sideband of the multiplex signal is not being used. In practice we did not find the S-MPX mode made much difference under our signal conditions, but it may be just the thing in your signal environment. One is left to wonder how well the circuit would work in a digital implementation, where the nonideal effects in the phase shifters and mixers of this analog implementation would not exist.

The ergonomics of this tuner are very good. Included are all the super-tuner goodies, such as a tuning knob with rotary shaft encoding and a signal indicator that gives signal levels in dBV. There is a separate IF strip, a detector circuit (another PA5008), plus an A/D converter to make this work. The LED display that gives the signal level also shows the preset station number. Why for $900 you cannot have separate LEDs for each function is beyond me. Other goodies include 10 kHz fine tuning (which can sometimes help under bad signal conditions by allowing the tuner to move away from a strong signal), variable and fixed line outputs, A/B antenna selection, and a tuning mode that allows direct entry of a station's frequency. The autoselect operation mode does not appear to do as well as that of the Onkyo T-9090II. It never went into MPX noise reduction or mono mode even when the signal clearly needed it. It also will not update automatically if signal conditions change, the way the Onkyo does. Oh, and I forgot to mention the F-93 does not have a remote control, although you can control it with other Pioneer equipment.

The F-93 has AM, which Richard Modafferi tested. He notes it has variable selectivity, and the noise figure is good. From his upstate New York site he could get Toronto and New York City. Richard has a station broadcasting at 1430 kHz 600 feet away. That wiped out everything above 1100 kHz. He notes that any 1950s vacuum-tube table radio would have no spuriae.

When you open up the F-93, you see one of the most complex tuners ever made. Construction is typical of mass-market Japanese equipment, but you can see $900's worth of stuff. Now, engineers have a principle called KISS (Keep It Simple, Stupid!) and the Pioneer F-93 violates KISS with a vengeance. The result is performance that is mediocre at best in normal operation or with strong signals. On the other hand, it can under some weak signal conditions deliver better reception than any other tuner currently available, and it does that for me. I actually bought this thing. But purchaser beware! Get a home trial before you use your credit card. The F-93 could do great things for you or it could make things worse. If you are using an outdoor antenna, or trying to receive a strong clean signal, then worse is more likely than better. Also note that this tuner has more adjustments than any other. At least 50—count them, fifty. Many of the adjustments interact, so it takes a long time to adjust this set, and adjustments will drift with time and temperature. What you get out of the box may not be properly adjusted. I asked Richard (a true believer in KISS) to tweak up my unit, but he declined. I was not surprised by his answer.

Rotel RT-990BX

Rotel of America, Equity International, Inc., 54 Concord Street, North Reading, MA 01864-2699. Voice: (508) 664-3820. Fax: (508) 664-4109. RT-990BX stereo FM tuner, $749.90. Tested sample on loan from manufacturer.

This is the lower-cost "stripped" version of the Rotel RHT-10. The folks at Rotel decided to send us one after they calmed down about the review we had given one of their power amps. The tuner is almost identical to the RHT-10. The high-blend function is gone. The signal-strength display is a set of 5 bars instead of the numerical readout of the RHT-10 (the separate meter amp is still included, only the display has changed). Board construction and layout are almost identical, with most parts on the board identical between the two units, including the critical stuff like the IF filters. Resistors remain metal film. Pioneer, Sony, Onkyo, and Yamaha use carbon resistors. The RHT-10 does have a better enclosure and a toroidal transformer, but nothing I saw would lead me to expect a difference in performance between the two tuners.

One nice feature of the RF section of both the RT-990BX and the RHT-10 was not explained correctly in my RHT-10 review in the last issue. I wrote that the RF section can be bypassed as in the Onkyo. In fact, it is not like the Onkyo. In the local mode the RF section remains active but another LC circuit is connected at the RF input. This triple-tuned filter improves selectivity at the RF input but at the cost of some insertion loss and hence reduced sensitivity.

In actual use, the RT-990BX turned out to be as good as, or better than, the RHT-10 in some respect but not others. On our killer 91.3 MHz test only slight splatter was heard from the 200 times larger signal at 91.5 MHz, provided the RF attenuation was engaged. Similar results occurred with the RHT-10. The RT-990BX cross-modulated badly on the 1 V signal in our 92.1 MHz killer test, and 91.7 MHz could not be received even with the attenuator on. The RHT-10 had no problems with the 1 V signal at 92.1 MHz and 91.7 MHz could be received. We also found serious $2f_1 \pm f_2$ spurious signals (f_1 and f_2 are 105.7 MHz and 105.1 MHz) at 106.3 MHz. The station at 105.7 MHz produces a 250 mV signal. Obviously the very weak 106.3 MHz could not be received, but in addition the signal at 106.1 MHz could not be received either. The attenuator again made no difference. Notes on the 106.1 MHz signal are not in the evaluation of the RHT-10, which probably indicates the RHT-10 received it. The signal at 105.3MHz (10 µV) was received on the RT-990BX. So cross-modulation from 105.7 MHz was not a problem. The tuner did get a 3 µV (75 miles away) signal at 88.3 MHz that was surrounded by adjacent-channel local signals, but crosstalk from the signal at 88.5 MHz was heard. These results were good, but why the RHT-10 did much better is a mystery. Perhaps the board layout changes (which would have to be very small, since the layout looks the same to me) caused some coupling that does not occur on the RHT-10. Perhaps an alignment problem in the RT-990BX caused the differences, although the tuner did not appear to be out of alignment. It could also be due to variations in the RF components between the two samples.

On audio performance things were reversed. Measured performance showed a 1 kHz THD of -72 dB, but recall that the Rotel RHT-10 did only -60 dB. Variability of the quadrature detector coil is responsible for these very different results. With age, drift will occur and that low distortion may not last. Even changes in temperature could change the results. The 10 kHz IM was also very good at -71dB. In the narrow mode the 1 kHz distortion was -45 dB and the IM distortion was -58 dB. Channel separation was 44 dB at 100 Hz, 55 dB at 1kHz, and 38 dB at 10 kHz. Narrow-mode numbers were only 3 dB worse, a very good result. The RHT-10 was less good in all these tests. The RT-990BX has the same very sophisticated MPX decoder as the RHT-10 except for a change of output op-amp. They went from an AD847 to an NE5534. No big deal, since the change, if anything, helped the performance of the RT-990BX. The high pilot tone of the RHT-10 was not present in this tuner. The canceler circuit must have been better adjusted. The output filter is still 2nd-order with no finite zeros. That will let a lot of subproduct out the end of the tuner.

The quadrature decoder in this sample of the RT-990BX is about the best we have seen but it is the luck of the draw, since our RHT-10 sample had worse performance. Contrast this to two samples of the Accuphase T-109. Both had near identical performance because a pulse-count demodulator was used. It requires no adjustment. It is impossible to tell whether the less good RF performance of the RT-990BX is due to component variations and alignment, or a systematic change from the RHT-10 in the RF stages' performance. Despite this, as only very large input signals give the RT-990BX problems, it still is a lot of tuner for the money provided you do not need some of the features found on other tuners (multiplex noise reduction, A/B antenna selection, tuning with a knob, AM, or a signal-strength meter calibrated in volts), and it is therefore highly recommended.

Sony ST-SA5ES

Sony Electronics, Inc., 1 Sony Drive, Park Ridge, NJ 07656. Voice: (201) 930-1000. Fax: (201) 930-4748. ST-SA5ES AM/FM stereo tuner, $800.00. Tested sample on loan from manufacturer.

Often readers will ask which companies produce the best products. Unfortunately it does not work that way. While some companies, such as Sony, produce more than their share of winners, and most high-end tweako companies can almost be guaranteed to produce products that are losers (at least from the engineering point of view), you still cannot tell what an individual component will do until you test it. So we had to test this top-of-the-line Sony tuner to see if it is a winner or not.

The ST-SA5ES tuner is similar to the 10-year old ST-S700ES. New features such as A/B antenna switching and rotary knob tuning have been added. Retained is the IF stage, which uses two ceramic filters in the wide mode with two more added in the narrow mode. Two of the IF amps are µPCl 163 ICs. The remaining stages are discrete circuits. Separate phase-tweaking coils are provided for mono and stereo operation.

Retained is the PLL FM demodulator. The loop bandwidth is changed in the narrow mode, trading distortion performance for better capture of weaker signals. The VCO is a surprisingly simple one-MOSFET affair that uses a single varactor. It is surprising that this VCO can have good linearity when it is not balanced. In addition to the VCO, a standard diode bridge mixer is used as the phase detector. An M5220P op-amp is used in the loop filter and PLL output buffer. The PLL mixer is driven by a µPCl163H IF amp that in turn is driven by the IF amp/limiter in the Sanyo LA1135. This chip also forms the meter drive circuit. No separate meter-amp path is included. This is a surprise for such an expensive tuner.

The Sony CXD1064S multiplex decoder is retained form the earlier design. This chip looks identical to the state-of-the-art Sanyo LA3450. I strongly suspect it was a codesign between the two companies. Sony had the CXD1064S out first. Often, in these IC development arrangements, the chip cannot be sold in the open market until after some period in which the developers have exclusive rights to it. In the ST-SA5ES, a new hybrid lowpass filter module is added for improved subcarrier rejection.

What has been removed from the ST-SA5ES is the tracking bandpass filters in the RF section. In the older tuner these filters were being shifted to correspond to the instantaneous frequency of the incoming signal. It is unclear why this feature was removed. It may have been a manufacturability problem or an attempt to reduce cost. In any case, it left only four tuned elements. In front of the RF stage there is only a single tuned circuit. A double-tuned circuit precedes the mixer that is not balanced.

New in the ST-SA5ES is an RF attenuator. That is something required for a front end that has poor dynamic range. It did not help. The tuner failed all our difficult outdoor reception tests. Bad spuriae were observed. The signals at 91.7 MHz, 106.1 MHz, and 106.3 MHz discussed in the reviews above could not be received. The stations in our tough selectivity tests were also not receivable. Included in the selectivity tests were stations at 88.3 MHz, 89.5 MHz, and 91.3 MHz that have been discussed above. As already explained in this survey, poor RF performance prevents accurate assessment of a tuner's real-world selectivity since, by definition, a selectivity problem also involves signals with large dynamic-range differences.

Indoors things were better, but not much better. We could still not receive 91.3 MHz because 91.5 MHz was still too strong. Furthermore, 96.3 MHz, 106.9 MHz, and 107.1 MHz were noisy. Since 96.3 MHz is the 9th harmonic of the IF and 107.1 MHz is the 10th harmonic, this noise is an indication that the tuner tailbites. Tailbiting happens when harmonics generated in the IF (which has lots of harmonics because of all the limiting action) get back into the RF stage.

It is unlikely that the tuner was defective or missaligned, since our instrument-based tuner tests came out OK. In the wide mode, 1 kHz distortion was -67 dB and in the narrow mode -47 dB. The 10 kHz IM results were -71 dB and -54 dB for the wide and narrow modes, respectively. Channel separation is 40 dB or better in the wide mode and 34 dB or better in the narrow mode.

AM use test results showed good selectivity and quieting, but like the FM section the AM had poor performance with respect to spurious responses and cross-modulation. The tuner was able to receive 770 kHz (WABC, New York City) at a distance of 195 miles in Binghamton, NY, but above 1200 kHz everything was wiped out by the very strong local signal at 1430 kHz. The tuner did receive the strong signal well, however, indicating good AGC action. The AM selectivity switch also proved useful.

Obviously, Sony needs to reengineer the front end of their top-of-the-line tuner to improve dynamic range and eliminate the tailbiting before we can recommend it.

Yamaha TX-950

Yamaha Electronics Corporation, USA, 6660 Orangethorpe Avenue, Buena Park, CA 90620. Voice: (714) 522-9105. Fax: (714) 670-0108. TX-950 AM/FM stereo tuner, $429.00. Tested sample on loan from manufacturer.

It was not our intention to save the clear bargain in this survey until the end, but it so happens that this Yamaha tuner is it, and we do things alphabetically around here. At a mere $429 we have here a tuner that is clearly inferior only to the $2995 Accuphase T-109 and no other. The FM front end has a very wide dynamic range. Perhaps it should be no surprise that the topology of the front end is similar to that of the Accuphase. Double-tuned circuits are used before and after the RF stage. The mixer is not balanced. An RF attenuator can be engaged at the input, and with that function activated we had no cross-modulation or spuriae to interfere with reception of stations that could be received by this tuner.

The IF stage consists of discrete IF amplifiers whose complexity varies from one to six transistors. Two ceramic filters are used in wide mode, with another added into the IF section in narrow mode. Selectivity of the IF strip was good enough to receive 91.3 MHz with some audible splatter. The Accuphase and Onkyo have more selective filters and less splatter. The old Mcintosh MR-78 has still more selectivity and no splatter. The very weak signal at 106.3 MHz that is normally covered up with spuriae was still not receivable because the local signal at 106.5 MHz interfered as a result of inadequate selectivity. For the same reason the difficult stations at 88.3 MHz and 89.5 MHz also could not be received. Moving indoors produced closer to excellent results. All Pennsylvania torture tests were passed. In Binghamton, NY, performance was also good indoors, although lower adjacent-channel selectivity did prevent the tuner from receiving at least one station that those with greater selectivity could get.

The demodulator and stereo decoder also show excellent performance. In the wide mode the 1kHz distortion in stereo was -71 dB. That is a hairsbreadth away from the Accuphase. In the narrow mode the 1 kHz distortion rose to -61 dB. Recall that the unit's narrow mode is not as narrow as that of the best super tuners; thus these distortion numbers are better than in the narrow mode of those tuners. This is the reason why Onkyo has three IF modes. Stereo IM test results were also excellent (2 dB better than on the Accuphase!), with -76 dB in the wide mode and -68 dB in the narrow mode. Channel separation over the full band was 46 dB or better in the wide mode and 40 dB or better in the narrow mode, also very good results.

The excellent performance of this tuner is no doubt due in part to the Sanyo LA3450 multiplex decoder used in the design. The 4th-order passive filter that follows also helps by doing a good job suppressing subcarrier products. But the low distortion numbers also result because the FM demodulator is doing its job well. At this tuner's price you would think you could only get a quadrature detector, but it turns out that an even older technology, the ratio detector, is used!

Those old enough to run for president may recall all the ink that was spilled in discussing the advantages and disadvantages of the Foster-Seeley "discriminator" and the ratio detector. These two detectors, along with the quadrature detector, use similar methods to demodulate FM. A time-delay circuit approximates a differentiator that converts the frequency-modulated signal to one that is both frequency- and amplitude-modulated. An AM demodulator then recovers the signal. In the quadrature detector, the AM demodulator is a balanced synchronous detector. In the Foster-Seeley, it is a balanced envelope detector that uses a lot more passive components. The transition from detectors like the Foster-Seeley to the quadrature detector occurred because the latter detector requires fewer passive components, but it needs a mixer circuit that became easy to manufacture only with the onset of IC technology.

The ratio detector looks very similar to the Foster-Seeley detector—one diode is reversed and a large capacitor added. But those changes make a big difference. The ratio detector performs the function not only of an FM demodulator but also of a dynamic limiter. It has internal AM rejection unlike other circuits in the time-delay-differentiator/AM-demodulator category. The downside of the ratio detector is that it is not balanced. That should translate into more distortion but it does not in the case of the TX-950. Careful selection of the passives, a time-delay differentiator design that is accurate over wide frequency deviations, and a modified topology that uses an op-amp (NJM2068S) as differencing amp in the demodulator account for such an amazing performance in this classic circuit. Please note that the circuit works well only if it is precisely adjusted. It turned out that the Yamaha was one of the few tuners that did not need any tweaking to get it to perform well.

One more reason for the low distortion of the TX-950 is that it uses a high-tech form of automatic frequency control. The PLL frequency synthesis brings the station in. Then, once the station is acquired, control of the varactors in the front end of the tuner is switched over to the AFC system. Even if the incoming signal is slightly mistuned, the AFC system will still ensure that the IF signal is centered around 10.7 MHz, where minimum distortion occurs. The AFC system also has the potential to create a local oscillator signal with less phase noise because the control voltage to the varactor can be more heavily filtered. This could improve signal-to-noise ratios and reduce spurious interference.

The AFC system is not the same as found in your old '60s tuner. It is built with a totally separate IF strip and FM discriminator. Yamaha uses the Sanyo LA1266 AM/FM receiver chip to do this (they also use this for the AM section). Yamaha's AFC used to have the problem of switching between the two tuning systems on weak signals—at least my 10-year old Yamaha T-70 liked to do this. The TX-950 has a more advanced design that does not have this problem. The meter circuit of the TX-950 also uses the LA1266 to provide a wide range of signal-strength indications, just like the state-of-the-art tuners.

Despite its low price, all the high-end features are on the Yamaha. They include A/B antenna selection, a tuning knob that drives a shaft encoder, call-letter display option, 40 presets, etc. (One small omission is the absence of threads on the 75Ω antenna jacks; they accept only push-on connectors.) Even the AM tuner is good. No spuriae or overload occurred from the local 1430 kHz signal, and the weak signal from WQEW in New York City could be tuned in. Richard notes that the AM tuner section works as well as 1950s vintage six-tube radios. How comforting that modern electronics has made no progress in 40 years.

Recommendations:

The best production tuner is the **Accuphase T-109,** and at its price it had better be. The **Yamaha TX-950** comes quite close to the Accuphase at a much lower price. Lacking only the ultimate in selectivity (but achieving better audio performance in the narrow mode as a result), the Yamaha is clearly a best buy. Obviously, the build quality and appearance of the TX-950 are those of a typical Japanese audio component. The only tuner in this survey, other than the Accuphase, that uses higher-quality parts and can be recommended is the **Rotel RT-990BX.** That tuner also puts the emphasis on high-end audio-circuit design, if you care about such things. To achieve the low $750 price point and still use quality parts, Rotel had to leave something out, and what they left out were features. The **Onkyo T-9090II** is still the DX champ at a price under a thousand dollars, but this is an old design and the multiplex decoder does not perform as well as those of the other recommended units. The Onkyo's build quality is similar to the Yamaha's. The **Magnum Dynalab 205** 'Signal Sleuth' RF front end is also recommended for potentially improving weak signal reception when using good indoor antennas. •

David Rich on Yamaha AX-570 *(continued from page 54)*

this one application too expensive. Electrolytic capacitors are in the input, output, but not the more capacitor-sensitive feedback loop, of the line stage.

In between the line-stage output and the power-amp input lies the Pure Direct switch. With it engaged, we have completely described the signal path. The signal at the input plugs flows through the input selector to the volume control to the line amp and then into the power amp. If Direct is not engaged, the signal ends up going through a cheap balance control (with detent) and then on to the tone control stage (more cheap controls). If you want to put an active crossover between the preamp and the power amp, you will need to keep the Direct switch disengaged because the loop is not in the direct path. The Pre Out is active all the time to allow connection of a subwoofer in a system that does not need a highpass filter in the main signal path. All low-power circuits are powered from taps on the main transformer. A separate bridge rectifier drives 3300 μF of filtering capacitor. The regulator is in the high-end style, with a simple pass transistor driven from a zener diode reference.

Our measurements yielded the following results:

In the power amplifier section, THD + N reaches a minimum of -94 to -96 dB from 20 Hz to 20 kHz (meaning virtually no dynamic distortion) with a load of 8.15Ω. Clipping occurs at 115 watts. Into 4.17Ω, the 20 Hz to 20 kHz minima are in the -90 to -94 dB range, and clipping occurs at 200 watts. Channel separation, measured at 1 watt output, is 100 dB at the lowest frequencies, decreasing to 70 dB at 20 kHz. Most remarkably, the above figures barely change, certainly no more than a dB or two, when the preamp line stage is included in the loop—with the Direct switch engaged, of course. When the Direct switch is disengaged and all the controls are in the signal path, at least 12 dB of deterioration is observable.

The phono preamp uses a discrete differential pair driving a UPC4570 op-amp. Distortion and frequency response are respectable but would be better if two-gain-stage topologies and better passive components were used. Our measurements in the MM mode showed THD + N minima of -85 to -87 dB at maximum output (approximately 7 V at Tape Out) in the entire audio band from 20 Hz to 20 kHz. In the very high-gain MC mode, on the other hand, the minima were only in the -60 to -64 dB range. RIAA equalization accuracy was ±0.15 dB up to 10 kHz, but at 20 kHz there were inexplicable errors of -0.6 dB (left) and -0.3 dB (right).

Separate wafers on the function selector are used for each tape-monitor output. This prevents self-oscillations. The tape monitors are not buffered, but by setting the tape selector to a position other than the main selected program, you can ensure that the selected input will not be loaded.

The AX-570 comes with a remote control that also controls the CD player, tuner, and two tape decks, provided they are also by Yamaha. One remote instead of five. What a novel idea. The volume control and selector switch are motor-driven for remote operation. A semiconductor switch approach (B&K) offers more reliability but may cause distortion. Engineering involves a lot of tradeoffs, with no single "right" answer. In this case, the Yamaha engineers went for the low distortion figures.

Measured results and use tests of the AX-570 show that $499 will purchase as much electronics as most of us will ever need. That does not mean that spending more is not a rational decision, but it does mean that you are on the steep part of the curve of diminishing returns. Perhaps the only real design flaw is that the ventillation slits in the cover are large enough to allow a child to drop change inside the unit.

As for those of you who want to know "how it sounds"—well, we must not be getting our point across. •

Thrice Shy: Multichannel Music Formats Further Considered

By Daniel C. Sweeney, Ph.D.
Freelance Contributor to *The Audio Critic*

This is the second part of a critical examination and evaluation of mutichannel audio technology, with regard to both hardware and software, present and future, fact and hype.

In this, the second in a series dealing with the issues of multichannel music reproduction, I will examine what are perhaps the two key considerations for music lovers: (1) the adequacy of multichannel recording techniques in terms of what is known about localization by the human ear and (2) the fidelity of multichannel playback systems according to the same criterion. I will also have a few words to say concerning the use of music systems in video applications.

FIVE-CHANNEL RECORDING

Antecedents: the Two-Channel Models

Within the realm of stereo recording practice, several distinct approaches have been developed for deriving differential inputs from the wavefronts representing the performance itself and then distributing the information so obtained between two channels, so as to create spatial effects upon playback. Without being in any way exhaustive, one can cite various binaural techniques (arguably not stereo at all in strictest definition), M-S stereo recording (also known as intensity stereo), the crossed figure-eight method associated with Blumlein, angled cardioids, spaced omnis, and synthetic techniques whereby a multitude of microphones is assigned to a number of musical instruments and voices in any of an infinite number of arrangements. It should also be noted that the last technique or family of techniques, ubiquitous in popular recording and derisively dismissed by purists as multiple mono, can be combined with minimalist techniques so that the minimalist core engenders the overall spatial perspective, while the additional microphones provide accents, so to speak.

While cogent arguments have been persistently advanced to the effect that only the original Blumlein techniques constitute true stereo, such positions have retarded not at all the diversification of two-channel recording techniques, and today extreme pluralism characterizes musical recording practice for all forms of music. And, correspondingly, extreme pluralism also characterizes the playback array, with loudspeakers exhibiting all manner of directivity characteristics vying for consumer allegiance, and consumers themselves deploying such speakers in all manner of listening spaces.

With such extreme diversity existing at both ends of the signal chain within the two-channel realm, little in the way of precise matching of pickup to playback system is possible, though on a strictly rational level there is indeed something to be said for matching speakers to microphones—for example, playing spaced omnimiked recordings over omnidirectional loudspeakers, or crossed figure-eight Blumlein recordings over toed-in dipoles. Still, we enjoy satisfactory listening experiences despite this rampant heterogeneity, and we are accustomed to expect acceptable stereo without too close attention being paid to the establishment of correspondences between the recording setup and the playback system.

From Two to Many

Here the question must be asked, what does such heterogeneity within the two-channel universe portend for multichannel? Will similar centrifugal tendencies manifest themselves in the age of multichannel, and to what effect?

Damned good question, in fact the key question in the whole discussion, because it immediately gives rise to at least two other crucial questions. Just how do we use these extra channels when we can't come to any consensus on how to use two? And what happens when we try to juggle intensity and phase relationships among several channels?

Now one might suppose that with all the confidence being expressed within the hardware industry about the benefits of multichannel, definite answers to such questions would have been framed. But sadly the opposite is the case. The general level of insight on the problems and possibilities of multichannel is lower than was the case a quarter century ago when the abortive quad revolution was launched, and of all the many individuals I've interviewed on the subject over the last year, only two, John Eargle and Tom Holman, seemed to have steeped themselves thoroughly in the theoretical underpinnings of multichannel recording and to be proceeding on the basis of a well-defined model of human hearing. And, interestingly, both individuals were quick to admit that much remains to be learned regarding recording for multichannel playback.

In other words you, gentle reader, will be part of a grand experiment should you add channels to your music system at this early date. Multichannel recordists of various persuasions will present you with a vast melange of program material embodying a dizzying profusion of "takes" on the deployment of five channels, and you'll be left with the task of sorting it all out.

But lest this scenario appear unrelentingly grim, let me hasten to point out that multichannel boosters—who include nearly everyone on the hardware side—are quick to object that subjectively successful two-channel recordings have been made with any and all of the common techniques, and that therefore no reason can be adduced that five-channel formats should not prove equally adaptable.

On the surface that argument seems plausible enough, but ignored is any due consideration of the fundamental change that occurs when the listener is surrounded by loudspeakers rather than facing a pair of them. Ignored as well is the much greater precision afforded by five discrete channels over two, which is to say that the information density of the recording becomes correspondingly greater, and directional cues are brought into play that scarcely exist in a stereo system. And finally there is the weighty issue of multichannel pans, where phase and intensity differences are juggled to produce phantoms between adjacent pairs of speakers, or even steered into the interior listening space described by the circuit of speakers, rather than being confined to the periphery as is always the case with two-channel stereo.

What I'm saying is that we may find two-channel stereo acceptable precisely because it is a loose and lossy approximation of what we actually hear in an actual performance space, and further that this approximate nature of two-channel may be what permits the multiplicity of techniques. To cite an analogy, it is a far more difficult and exacting task to make a 3-D motion picture that provides an acceptable illusion of reality than it is to accomplish the same goal with conventional single-camera flat-perspective movie footage, the 3-D format imposing upon the film maker a myriad of highly frustrating constraints as a price of the added dimensionality. At least some of the available evidence suggests that multichannel may be similarly problematic and that five-channel may prove to be less than the unalloyed enhancement that industry spokesmen would have you believe. But I mention this only as a possibility. The fact is that we simply don't know with any certainty how five-channel will accommodate different recording techniques, because nobody has enough experience to speak definitively on the subject.

If this statement appears remarkable, consider the fact that a mere handful of recordings were specifically miked for multichannel back in the quad era. Almost everything released in quad was simply remixed from the same multitrack master used for the stereo release, with a quad pan-pot being used in an attempt to place sounds at intermediate positions between speakers. True, during the last few years, quite a bit of mixing has been done for movie formats utilizing five full-range channels, but the recording of motion-picture sound tracks diverges in both its aims and its techniques from the modes that define purist music recording. Movie sound tracks place their stress on arbitrary effects, not on the recreation of a specific performance event, and, moreover, film sound is characterized by extremely layered compositions far exceeding most multichannel pop recordings in density and complexity. Thus it is fair to say that, in regard to music, the art of recording for multichannel is still in an embryonic state of development.

I should mention here that in the researching of another article on this same general subject I went to great lengths to interview recording engineers with a clear commitment to multichannel playback and with extensive experience in the same. Unfortunately, I found that only a small minority of recordists have a clear commitment, and almost no one has much experience, John Eargle and Brad Miller being just about the only individuals who can really be considered veterans. Lots of people have experimented at one time or another, but these two individuals are the only true gurus, and, since neither is regularly employed by major labels anymore, you won't get the benefit of their experience—if 5.1 or some other multichannel standard ever does become the dominant music format.

Bedrock: the Sampling of Acoustic Space with Two or with Multiple Channels

A sense of space is created on a recording principally by two means: by capturing localization cues that enable listeners to identify the positions of instruments and voices within a recreated acoustic space, and by capturing the pattern of reflections and reverberation that conveys the hall sound (even to the approximate dimensions of the recording venue). Bear in mind that these two perceived aspects of spatiality are not at all the same thing and should be clearly distinguished in discussions as to how an audio system recreates the space of a performance, though, unfortunately, they are often confused in subjective discussions of loudspeaker imaging.

First let's discuss sound localization.

I will assume that most individuals reading this journal are aware that interaural arrival-time differences and interaural amplitude differences constitute the most significant localization cues for humans (if you're unfamiliar with terms, consult any standard text on recording). Although most members of the audio press appear to remain ignorant of the fact, there is a large body of evidence for the existence of a third important localization cue known as the HRTF (head related transfer function), a psychoacoustic phenomenon resulting from the differential diffractive effects imposed by the structures of the head and outer ears on sounds arriving at varying angles of incidence. Less conclusive evidence may be cited for the existence of one other major cue arising from interaural phase differences.

Several other localization cues have been posited as well, but the above mentioned three are surely primary, with phase differences occupying an insecure and ambiguous position.

In stereo recording, amplitude cues are stressed, and often they are the only cues provided; however, among the classic techniques, spaced omnis and angled cardioids capture arrival time differences effectively as well. Interestingly, coincidental techniques such as crossed figure eights or M-S normally do not capture arrival-time differences. Significantly, no microphone techniques other than binaural techniques utilizing dummy heads with dummy ears can capture the HRTFs, though these can be synthesized by appropriate postequalization.

Now what about recording for multichannel playback? Do the recording techniques devised for the new media improve upon the situation and provide full employment for all localization cues?

The answer to that is a qualified no, and to understand why this is the case one must realize that for all the hoopla surrounding 5.1, relatively little research has been devoted to tapping its possibilities. (Here it must be said that various individuals and organizations such as the ARA [Acoustic Renaissance for Audio], Tom Holman, and John Eargle have proposed an open standard that would allow for selectable channel number as well as variable bit and sampling rates.)

In fact, no really new recording techniques have yet been developed for 5.1, though John Eargle has suggested some of the directions in which recordists might proceed. The two individuals who have been most active in 5.1 music recording, Brad Miller of Mobile Fidelity International and Tom Jung of *dmp*, both use techniques dating from the quad era, and it is safe to say that recording for multichannel, such as it is, represents no significant advance over the quad experiments. People are talking about a revolution, but the basic recording technology is stuck in the '70s.

And just how did our distant ancestors back in the '70s record for four-channel playback?

Several formulae are mentioned in the literature, including three-channel matrix techniques which I will disregard here as having no direct application, but basically two approaches obtained.

The first approach, owing much to purist stereo techniques, was to use a fairly conventional minimalist miking setup such as angled cardioids up front and then to place another pair of mikes—often widely spaced omnis—in the rear of the hall. A variant was the use of cardioids in the rear, facing away from the back wall. Most of the purists thought that rear channels should be chiefly used for reproducing ambience, and some even suggested recording left rear and right rear out of phase to inhibit the formation of a rear phantom image.

The second approach was an extension of the pan-potted multitrack methods which had become the norm in popular recording by 1970. As in stereo multitrack recording, each instrument and voice was assigned a separate track, and performers might even be recorded apart from one another in booths or gobos. The performance, which could be entirely synthetic, was essentially assembled during mixdown, and performers were distributed about a synthesized acoustic space via the mixing console and the pan-pot. Extending the approach to quad meant simply mixing down to four rather than two channels and involved similarly arbitrary decisions as to what went where.

The first approach, building on classic stereo recording techniques, was explored by many eminent audio professionals, including Eargle, F. Alton Everest, Peter Scheiber, and E. Roerback Madsen, but it chiefly found expression in their published ruminations rather than in actual recordings. Later though, at the close of the decade of the '70s, one particularly dogmatic form of the purist approach represented by the Ambisonics camp did result in actual recordings, albeit only a few dozen altogether. Unhappily, the cult tendencies and tendentious proselytizing of its advocates served to marginalize the Ambisonics format, and today it seems unlikely to exert much influence on five-channel music recordings. Nevertheless, the purist approach lives on in Dolby matrix recordings put out on the Delos label and in the Mobile Fidelity International recordings engineered by Brad Miller, and it is fairly certain to see some future applications at least in the realm of audiophile recordings.

The second approach, i.e., the synthetic approach, resulted in recordings which, by and large, significantly detracted from the credibility of multichannel. Rear channels were given equal weight to those in front, and placement of instruments in back of the listening space was frequently sought. Circumferential dynamic pans were also attempted. This type of treatment was the norm in the hundreds of quad remixes of popular releases.

During quad's heyday, such as it was, these aggressive surround mixes invariably came to naught due to the very poor rear-channel separation of the dominant SQ format, and today quad defenders are quick to marshal the low-separation argument to explain all of the failings of quad as a popular medium. Certainly low separation effectively thwarted any attempt to place sources at points beyond the frontal soundstage, and yet I believe that limitations of the low-separation matrices were not the only factor in the mass rejection of quad in popular music. The real problem lay with the listener.

If current scientific theories of human localization are even remotely correct, the interaural amplitude, arrival time, and phase differences only permit pans in front or in back of the listener. Images cannot be panned between a front left and rear left speaker or a front right and rear right speaker because, in the case of amplitude differences, the opposite ear is completely shadowed for all positions, while arrival time and phase differences vary by too small a degree to produce stable phantoms. Furthermore, in the case of pans behind the listener between rear left and rear right speakers, the precision of phantom localization appears to be far less than in the front quadrant. Simply put, four- and five-channel formats, used like pan-potted stereo, demand a kind of listener who doesn't exist, a listener whose localization abilities are uniform throughout the horizontal plane.

If humans indeed lack the ability to form stereo phantoms in all quadrants, then the only option open to the recordist who wishes to emphasize the rear channels and provide the multichannel convert with his money's worth is to localize sounds at the rear speaker locations—hard left back and hard right back. And that will give us the same hole-in-the-middle effects and ping-pong lateralization that was popular in the earliest days of stereo. None of which augurs well for a multichannel renaissance in the field of popular music.

Not unexpectedly, most marketing people in the hardware industry grow very cross when presented with such information because for them pronounced back-channel effects are crucial in the promotion of the format. Indeed I've been told by such types that I'm disloyal to the industry for even presuming to mention scientific studies that produce these negative findings, and that even if the studies are accurate, DSP will solve the problem. But it seems to me that all the industry loyalty in the world isn't going to change the nature of human perception, and if pop-music producers attempt to make recordings that flout the laws of psychoacoustics, as they did in the '70s, a successful reintroduction of multichannel becomes unlikely. And as for DSP solving the problem, the problem lies in the way human beings construct a soundfield from a finite number of channels. What they're asking for is a software fix for a wetware (neurological) limitation.

And that human limitation is worth examining.

What normally happens when one attempts to pan in every quadrant is that the soundfield becomes highly unstable, with the formation of inside-the-head phantoms similar to those experienced in headphone listening and sounds seeming to pop up in unpredictable places like shells falling on a battle field. The effect is interesting enough on first listen, but whether it will continue to beguile even the mass of unsophisticated consumers is questionable. It has been said that no one has ever lost money by underestimating the taste of the American public, and here we have an excellent test case.

To Market, to Market...

Obviously I feel very strongly that the aggressive use of rear channels in pop recordings will result in crassly gimmicked presentations, with little appeal to serious listeners. But of course serious and sophisticated listeners form a tiny minority, while the Beavis and Butt-head sensibility is overwhelmingly dominant. Knowing this, five-channel boosters suggest that gimmicked, ping-pong presentations will not only prove satisfactory but will eventually be insisted upon by the great American listening public.

Perhaps they're right, and such multichannel monstrosities will become the norm in music reproduction. But in the sunny pronunciamentos of the 5.1 camp I see several key questions being begged.

One of those questions has to do with the way listeners will set up systems in their home. Multichannel playback systems are not forgiving of haphazard placement, and the familiar consumer response to stereo—put one speaker behind a sofa and the other on a bookshelf—is not going to provide much of anything when extended to five full-range channels. Of course today, if industry sales figures are to be trusted, most of the new audio systems going out the door are already multichannel combinations, albeit designed for Dolby matrix playback, but no one knows exactly how these systems are set up in actual use or how many such systems are really operating at anything close to their potential. How many people, for instance, actually connect surround speakers to their Dolby Pro Logic AV receivers? And of those that do, how many have the speakers where they can produce a surround effect? Clearly such questions demand answers because if 5.1 is presented on systems consisting of three front speakers around the fireplace and a couple of eight-inch-high rear speakers somewhere on the floor in the back of the room, the results will do little to justify the overheated hype being generated by the industry.

Other unanswered questions pertain to marketing considerations—considerations which may ultimately be more important.

In introducing any multichannel playback format, one inevitably has to deal with the matter of backward compatibility, always a major concern to record producers in the past. Ensuring full stereo and mono compatibility in a five-channel mix isn't easy, and yet radio stations, which are not equipped to transmit 5.1 and may never be, are not going to stand for incompatible mixes.

There's also the issue of industry support on the software level. The fact is that no prominent recording artist or music producer has come out in support of 5.1 for music, and no major label has said a word about issuing music recordings in that format. The support is coming strictly from the hardware side and from a few individuals working for small audiophile labels.

Hardware supporters of 5.1 counter by arguing that five-channel music recordings can simply ride on the coattails of video, and they point out, correctly, that 5.1 is already available on laser disc. But that's a weak argument because laser disc itself is a format with negligible penetration after more than fifteen years on the market. The popular-music recording industry is not going to content itself with addressing the universe of laser disc supporters. The numbers just aren't there. For 5.1 music recording to piggyback on video, either HDTV (which will support discrete multichannel) is going to have to come on a lot faster than anyone expects, or the as yet unreleased *[now just barely released—Ed.]* DVD (digital versatile disc) is going to have to succeed magnificently.

Most 5.1 boosters in fact pin their hopes on DVD, but its success cannot be considered a foregone conclusion—witness the failure of Philips's current 5¼-inch digital video disc, which bears many points in common with DVD, industry protests notwithstanding. And, really, why should either small-disc video format succeed? The initial target market of early adopters already has a video disc, the venerable 12-inch laser disc, and it already has 5.1 on the 12-inch format. Why then should such buyers opt for a new format and one that is burdened with the uncomfortable issue of video compression to boot? And why should the specialty retailers who have loyally supported laser disc for two decades come out in favor of something designed to obsolesce the older format instantly and put them in ruinous competition with mass merchandisers, who are the primary target for DVD?

Representatives of the Japanese majors blithely brush such objections aside by saying that there's plenty of room for multiple formats in the specialty market, but the history of the industry has in every instance proven otherwise. What this means in terms of marketing strategy is that the DVD forces are going to have go straight to the mass market, push the hell out of the thing, and somehow convince all the video store owners to get into dual inventory or perhaps just scrap tape altogether. And maybe that will happen. Maybe the Japanese and the software manufacturers will initiate the aggressive—no, giveaway—pricing necessary to gain the consumer critical mass quickly, though they've certainly never done so with any previous format. Anyone want to pay me to do a video-software retailer survey on the subject?

In all the years I've been writing about consumer electronics, I've never encountered a tougher call than 5.1. I wish I could predict its fate with certainty because I think the response of the music lover ultimately has to hinge on market considerations involving this new format. If DVD fails, then where will 5.1 be? If Delos, *dmp*, and the like are the only ones who are going to be issuing 5.1 music recordings, are you prepared to revamp your music system? And what if some of the audiophile labels go for the Dolby AC-3 version and others for DTS? That's already happening now, with John Eargle favoring Dolby, and Brad Miller and Tom Jung solidly behind DTS.

Five-Channel Vaporware?

Might we now stand back for a moment and attempt to summarize the situation at the software end?

Recording techniques have not solidified, nor is past recording practice for multichannel generally in accord with what is currently understood about human localization capabilities. And even where recording techniques are informed by present-day science, the resources available to the five-channel recordist appear to be insufficient to map a three-dimensional acoustic space with total verisimilitude.

Correspondence between pickup and playback systems has scarcely been investigated systematically at all, either during the quad heyday or at the present. It goes without saying that detailed recording/playback standards are not in place—except in the case of the nonstarter Ambisonics format. (Ambisonics is being heavily promoted by the ARA group as the recording standard for the proposed audio-only high-density disc; my bet is it won't fly, but that's another article.)

Finally, a well-organized marketing campaign is not in evidence. Instead, 5.1 enthusiasts have simply assumed the inevitability of multichannel dominance and have not gone to the trouble of enlisting music-industry support, nor of educating the public beyond placing puff pieces in a few audio buff books.

Compared to the cogent, disciplined, and energetic promotional activities of the Compact Disc Group, the efforts of the 5.1 camp have been amazingly weak. Such slackness speaks highly of the latter group's confidence but does not reflect historical realities. During the last fifteen years nearly twenty new consumer software formats have appeared, but, of those, only a couple have succeeded, suggesting that concerted marketing efforts are well-nigh indispensable.

In sum, the software situation of multichannel music playback is disorganized in the extreme, at least as disorganized as the quad revolution was twenty-five years ago. True, in quad the format rivalry was infinitely worse, though SQ was heavily dominant, but, on the other hand, software support was certainly there along with a powerful publicity campaign. If quad failed with the full resources of both the software and hardware manufacturers behind it, what are the chances of 5.1 with admittedly better technical specifications but with little marketing impetus? I don't know.

My guess is that a real conversion from stereo to multichannel is not in the offing as regards music programming. In the longer term—say, five years from now—it may occur, but absent a coordinated marketing campaign I can't see how it can happen now.

Of course, I could be wrong. Of the two runaway format successes of the last fifteen years, Dolby Pro Logic did in fact succeed without the disciplined promotional efforts that marked the progress of the other winner, the compact disc. But Dolby encoding was already present on most stereo movie software anyway and was fully backwards compatible, and thus Pro Logic could be promoted successfully from the hardware side alone, since the software was already taken care of. A similar situation does not obtain with 5.1.

We shall see. My advice is to ponder long and hard before rethinking your music system for five discrete channels. It may be a long time coming.

THE PLAYBACK CHAIN

Even if 5.1 remains a minority video-only format—a distinct possibility—the enthusiast listener is still confronted with the dominance of matrix surround throughout the video realm and with the decision of whether to attempt to use a single system for stereo music listening as well as for home-theater sound reinforcement.

Much ink has been spilt in blathering on the suitability or lack thereof of audiophile speakers for movie sound and, conversely, home-theater rigs for music listening, and indeed the contention has grown rather ugly of late, with old-line audiophiles dismissing the home-theater crowd as tin-eared Philistines, while the latter mark the former as irrelevant fogies. And while one might wish to stand above the fray, doing so becomes difficult in a journal whose allegiance is clearly to the cause of promoting serious music. Still one can attempt to avoid the acrimony and to concentrate on the design issues, which should be the focus of the discussion.

Can One System Do Both?

To begin to answer the questions concerning the efficacy of dual-use systems, let us pose a hypothetical that surely applies to many of our readers.

Let us suppose that you as a music lover already own a high-priced, well-reviewed pair of stereo loudspeakers. It could be a pair of Avalon Ascents, or Quad ESL-63's, or Infinity Epsilons, to name just a few currently popular contenders embodying various design models. Let us further assume that you have a large sum of money in amplification. How might you retrofit this system for video applications, or need you retrofit it?

Or might a better strategy be simply to build an entirely separate system for video? Or just sell off that audiophile relic and buy an ambitious dual-use system, such as the JBL Synthesis or the Snell THX Reference?

Or is a retrofit even necessary when an audiophile system is pressed into service as a video PA?

Certainly, you can play stereo video material in straight two-channel form instead of decoding the center and surround channels, and that sort of makeshift would relieve you of the task of making modifications in your existing system. But, at the very least, such a strategy would rob you of surround effects and would compromise lateral pans across the front for all but the listener in the sweet spot. In my view, an optimal dual-use system perforce must include multiple speakers.

That being the case, could such a multispeaker system be built using an existing stereo music system as its core? Undoubtedly something could be done along those lines, but I believe that the results would be less than ideal.

To understand why this might be so, one must examine three areas: speaker matching, program requirements, and room interactions.

In terms of speaker matching, the user faces two distinct sets of problems— integrating a center speaker with the stereo pair and matching the surround speakers to all three front speakers.

Most people who've seriously investigated multichannel playback agree that the frontal array takes precedence, and that matching of front speakers is of paramount importance in achieving credible soundstaging. Indeed, listening experiments indicate that when the center speaker is not timbrally matched to left and right, pans—both static and dynamic—are seriously degraded. On this basis, center-channel augmentation would appear to require a single speaker of the same model designation as those used for stereo, or else something that is very closely matched in timbre.

Obviously, with mirror-imaged music speaker systems you're screwed. There's simply no easy way to add a center channel with the right characteristics. But even when both speakers are the same and the manufacturer is willing to sell you a single speaker, you may have problems.

The problems arise both from the differing natures of multichannel movie sound and stereo music programming, and from the loudspeaker directivity characteristics best suited to either application.

What follows here is controversial but cannot be avoided on that ground, as it is central to the whole discussion.

Stereo music software, particularly the purist recordings of acoustical music which we cherish, contains three basic types of spatial information: the direct sound at two locations represented by the stereo microphone pair, the early reflections from the front of the hall (generally captured by the same pair of microphones), and the diffuse reverberant soundfield from the back of the hall, which often is picked up by auxiliary ambience mikes. In consort these three types of spatial information suggest both the dimensions of the acoustical space and the placement of performers within it.

In two-channel stereo all of this information is folded together into two channels, where it is imperfectly decoded by the ear-brain. On the other hand, in film recording ambience is allotted to the surround channel, while early reflections are generally not an issue, due to the synthetic nature of the recordings. In five-channel music recording the channel allocation of early reflections is anyone's guess.

In actual two-channel playback in the home, the natural ambience in the music recording is usually supplemented by room reflections stimulated by the off-axis output of the loudspeakers themselves. The alternative, the LEDE (live-end-dead-end) concept whereby early refections from the front of the listening room are suppressed and supplemental simulated ambience is provided by rear-wall diffusion, has much to recommend it from a theoretical perspective but has never gained much popularity outside of the recording-studio milieu because of the expensive and ungainly room treatment accessories necessary to achieve it. (One might add that the speakers deemed appropriate for LEDE applications are such as are scarcely to be found in the consumer marketplace—combining, as they must, phase linearity with narrow directivity characteristics; the old Win SM-10 fits the bill nicely, but nothing else of which I am aware.)

Here I must point out that, strictly speaking, the front-wall, side-wall, floor, and ceiling reflections engendered by a wide dispersion stereo pair are not really analogous to the diffuse soundfield one experiences in a large listening space because of the uneven temporal and spatial distribution of acoustic energy. But in subjective terms the two are sufficiently alike to create favorable reactions among the majority of sophisticated listeners. Within a normal listening environment, wide-dispersion stereo speakers are generally deemed more natural-sounding than their beamier counterparts, though they cannot equal the imaging precision of narrow-directivity designs. There's also evidence that the acoustical cross-talk produced by wide-dispersion speakers is conducive to the creation of a more stable phantom center.

In the two-channel stereo mode, where such matters have been extensively studied, the off-axis output of the loudspeaker exerts several significant effects in addition to affecting apparent source placement and spatial perspective. Off-axis output influences listener perceptions of overall tonal balance, determines the width of the stereo window, and, in the lower frequencies, affects the coupling of the speaker with the room and the incidence of room modes and wall resonances. Furthermore, for reasons we needn't examine here, optimization of directivity characteristics for one purpose, such as achieving a fairly uniform perceived frequency response in varying domestic listening environments, is almost always had at the cost of compromised performance in other areas, hence the endless unresolvable debates on the relative merits of dipoles, bipoles, monopoles, and omnis. In toto, off-axis response virtually defines the subjective character of a loudspeaker and for that reason is subject to the most careful attention from competent loudspeaker designers.

Now, in multichannel systems the issue of directivity becomes vastly more complicated (as if it weren't complicated enough) due to the fact that ambience is chiefly assigned to side or rear speakers, a fact which, according to some, eliminates the need for or desirability of wide-dispersion speakers up front. Indeed, according to the folks at THX, who have probably researched the subject more extensively than anyone else, wide-dispersion designs for the front speakers are to be positively avoided in a multichannel playback system.

So what happens to all those carefully crafted omnis, dipoles, and bipoles which have found such favor with audiophiles in the past? According to Tony Grimani of Lucasfilms, such designs are apt to wane in acceptance as more and more customers warm to the idea of dual-use systems. In other words, you can consign your Quad ESL-63's to the dustbin of history. Instead you're enjoined to embrace the THX concept for front speakers, which is essentially that of a short line source—albeit only an approximate one.

And what happens when you use such narrow-directivity speakers for stereo music listening? That's where opinions differ. Those in THX fold, which includes most of the mainstream audio press, assert that THX speakers surpass conventional designs in all applications, but there are numerous dissenters, including many members of the subjective press. One can certainly argue that narrow-directivity designs, by avoiding floor and ceiling bounce, are inherently capable of better fidelity, but that isn't going to make people love them. The fact is that THX speakers sound identifiably different than the quasi-point-source radiators that comprise the bulk of the well-regarded audiophile music speakers, and that, I think, accounts for the rejection of THX designs in the audiophile press.

I find it interesting in this regard that JBL, in designing their supposedly state-of-the-art Synthesis systems, endowed the front speakers with dual driver arrays—one set for video and one for music—and that the directivity characteristics of the two arrays differed markedly. And the fact that Floyd Toole has been involved in the design of the Synthesis series gives us reason to believe that the notion of different directivity characteristics for different program material is not altogether unfounded.

So what does all this tell us about the feasibility of dual-use systems?

I have already indicated the difficulties of modifying a traditional music system for video use. As for scrapping the music system and going for a designated dual-use system, such as any of the THX-certified systems, I'd say audition the dual-use system carefully on music before you do so. I've never heard a video system that I considered an ultimate music system, but that's a personal opinion. Most of my peers would disagree. I would note, however, that video-oriented speakers, aside from having directional characteristics that are arguably ill-suited for two-.channel reproduction, never seem to approach the build quality of the best audiophile speakers. Really premium drivers are not used, cabinet quality is so-so, and crossover components are not of the highest quality. Of course, there are those who will argue that top-quality drivers and inert cabinets are needless extravagances, and that no one can hear the difference between a carefully designed system using average components and one where cost is no object. But then, if you believed that, you'd have never bought audiophile speakers in the first place.

Terra Incognita: the Anterior Portion of the Listening Space

Now what about the rear speakers?

For some reason the industry has fixated on the rear (Freudian overtones here?), and both speaker manufacturers and the audio press have become embroiled in two related disputes—whether the rear speakers should be of the same type as the fronts, and whether the rears should produce an output characterized by a preponderance of direct or diffuse sound.

Lucasfilms THX has taken a very clear and carefully reasoned position on the matter, advocating diffuse dipolar rear speakers having directional characteristics that are obviously different from those of the front triad. On the other hand, multichannel advocates with roots in the quad era, such as Gary Reber (editor of *Widescreen Review)* and Brad Miller of Mobile Fidelity International, insist that like speakers be used in all positions. Certain prominent high-end speaker designers, most notably Jim Thiel, also support the notion of like speakers all around.

While the dispute may appear to represent a classic clash of reason against mysticism, pitting the modern, scientifically informed THX gang against an unlikely alliance of fuzzy-minded high-enders and quad-era relics, in fact the roots of the controversy go all the way back to the earliest days of quad, and the issues are nowhere near as simple as the above construction might imply.

Most early researchers of multichannel playback assumed that, as a matter of course, like speakers would be used all around. Quad was seen as an extension of two-channel stereo, and like speakers appeared desirable and even necessary. And this attitude was not confined to scientific ignoramuses by any means. The mathematically minded prophets of Ambisonics always assumed that like speakers would be used.

However, within the laboratories of those two stalwarts of the Boston sound, AR and Advent, experiments were conducted in which omnidirectional rear speakers were used along with conventional monopole front speakers. Whether Tom Holman, the architect of THX, was directly aware of those experiments I can't say, but living where he did at the time he must have absorbed the notion that rear-channel output should be diffuse, a notion which found further expression in the a/d/s delay-derived rear channel, which often was assigned to small bookshelf speakers supplementing a main pair of towers in the front.

So who's right?

If matrix surround were all we had, diffusive rear speakers would clearly hold the advantage, despite the fact that the reverberant characteristics of a large hall can only be approximated in the home. But with 5.1 we have the capability of isolating sounds in left and right locations toward the rear of the listening space. Wouldn't more conventional speakers be more appropriate in that instance?

Lucasfilms says no, but it seems to me here that their usual rigorous logic is less in evidence here. If THX rear speakers are specifically designed to thwart localization to the speaker position, which they certainly are, how in the world can they reproduce sources that are specifically intended to be localized at the speaker positions? Mr. Grimani suggested to me that, after initial experimentation, mixers will end up using the rear channels pretty much as they always have, eliminating the conflict, but I tend to think that the very presence of split rears in the format is always going to encourage their use. They simply add an additional element of razzle-dazzle.

But to return to our discussion of music systems, should one then attempt to supplement one's expensive pair of stereo speakers by adding another pair of the same make and model in the back of the listening room?

I am certain that most of the manufacturers of expensive stereo music speakers would heartily encourage you to do just that, but I am not certain I can recommend that approach as either cost-effective or just plain effective. Experimental evidence suggests that there is indeed a problem of localizing even diffuse back-channel information at the rear speaker locations, which of course conduces to an unconvincing simulation of ambience. If this is the case, why then would you want to pay an additional five to fifteen thousand dollars for another pair of premium speakers which ultimately won't give you what you want? Especially when you can simulate rear-channel ambience by much less expensive means, by investing in a set of diffusive room-treatment devices from such manufacturers as RPG, ASC, or Systems Development Group. (Such devices will work with any speakers

(continued on page 74)

Two-Channel Stereo Is As Dead As the LP

By Tom Nousaine
(The High-Definition Weasel)

Major improvements in any performance-based technology, especially audio, come in quantum leaps. Fundamental leaps for communications were made possible by major advancements such as negative feedback, stereo, solid state, microelectronics, and digital storage formats. The architecture of the Home Theater system popularized by Dolby Pro Logic programs and high-quality multichannel decoders like the Lexicon have laid the basic foundation for the next major leap forward: discrete multichannel recordings. DVD is right around the corner.

Many folks will recoil at the suggestion of new formats because it suggests trashing all their existing equipment. Not true. Multichannel will be phased into existing systems just like stereo recordings and CD players. Some stuff eventually gets dumpstered (like my record-playing equipment), but most of it would be turned over eventually anyway. Just because.

So what's so special about multichannel?

First, Home Theater, with its basic multichannel architecture, has broken the "stereo as motorcycle" model. Second, the multichannel architecture brings better system implementation and setup, which improve even two-channel performance; and finally, it can be seen that the new format follows basic laws of natural progression in playback improvement. In between breakthrough developments, progress is seen through progressive changes that mainly reduce the price and improve the functionality and reliability of products. CD players, for example, have become much less expensive and sprouted improved functions like remote control and multiple play. They still all sound fantastic, although recordings have gotten much better because they are now recorded, mastered, and mixed for the new format. The first CD recordings were just reissues of stuff made for LP format release.

All you bikers know two-channel stereo is much like a motorcycle. You can have a passenger on a motorcycle, but only one person really gets the full experience. Stereo is a one-person party. Sweet-spot listening is the only way a stereo system can reach full potential. Period! Furthermore, using a top-quality stereo system is not a trivial task—how many of us have Do Not Touch and No User Serviceable Parts signs plastered all over our precious equipment? How many have family members who can actually use it? How many have families that are afraid to use it? How many have secreted the stereo in a separate room just to protect it from the prying fingers of family members? How many have family members who resent the money spent on Dad's toy? Most, or maybe all, of us. The parallels with the motorcycle are painfully obvious.

Home Theater is different. The system is installed in the family room, where style isn't an issue. Everybody uses it. Family members, even the wife, encourage resource deployment for Home Theater systems. You can even get home builders to include some of the basics in the price of the house!

Home Theater systems also bring important performance advantages with them because of their visual screen-based nature. First off, everybody agrees that a bigger picture is better. Bigger pictures mean bigger screens, which takes the heat off speaker size. A 50-inch tower speaker overwhelms the average living room but looks perfectly natural flanking a 50-inch rear-projection TV. Speakers are usually much more effectively installed in a Home Theater. The center channel is always on top of or under the screen, which gives new meaning to left and right. Because the TV sticks out into the room, speakers tend to be installed away from the rear wall in Home Theater systems—all without complaints from the family style consultant.

Listeners also become better placed. Because the system has a visual aspect, listening seats are dragged out into the room so viewers can see the screen. This means all listeners are facing the primary sound sources and may even be away from a nearby wall. Can you think of a time in the living room when putting a speaker or a chair out into the room wasn't contested? And—the big improvement comes with having multiple listening seats, all with an acceptable image perspective. No more herding special guests into the sweet spot. No more grappling or fighting for the good seat.

The separate subwoofer architecture also brings performance advantages. Placing a full-bandwidth loudspeaker in locations that provide optimal spectral and spatial rendition compromises bass in most rooms. Because room modes are widely spaced at low frequencies, the best stereo locations also deliver a big hole in response somewhere between 30 and 50 Hz. For example in my room (12 by 22.25 by 8 feet) the best left, center, and right locations leave a big suckout at 35 Hz because the two lowest room modes occur at 25 Hz and 47 Hz. The fix is to put the subwoofer in a closed corner, where it excites enough multiple wall modes to fill in the hole at the main listening positions. Without a separate subwoofer you cannot position both the main channels and the bass optimally.

Finally, to get a major step function in performance we need *more* than two channels. It can easily be seen that two channels are better than monaural. Most of us have forgotten that stereo, as invented at Bell Labs in the '30s, was a three-channel format. Home Theater gave us the third channel back. What's the next natural progression? Five channels. Then ten. The multichannel formats now available use five channels. So will DVD. Five channels are a major step forward in realism in the living room.

The move to five channels is far more important than all the sampling-rate and bandwidth haggles going on, such as 96 kHz with 2 channels—no better than 44.1 kHz and stereo. DVD will use 48 kHz, which is more than adequate. Do we need more bits? Not for a media release. Sixteen has been more than adequate for over a decade now. More bits may be needed for production but not for consumer release formats. Lossless coding? Not an issue. Data reduction? No problem. What we need is a minimum of five playback channels. Five discrete playback channels provide a solid, deep frontal image that remains stable for a number of listening positions, allows listener head movement, and fills the room with hall ambience that envelops the listener instead of clustering around the front channels.

Multichannel is the next big thing. It provides a major step up in playback performance and realism. The first programs will come from the movie people, as always. They dragged us into stereo in the '50s, matrix surround in the '80s, and are now dragging us into discrete multichannel with DTS and Dolby Digital. The established recording industry can be expected to drag their feet. Most of the several thousand extant recording studios are already at capacity making two-channel recordings, jingles, and commercials. They will see no reason to change anything until the multichannel recordings being made on new digital workstations steal enough of their business to create excess capacity. When that juncture is reached, you will see a wholesale changeout.

How long will that take? Well, in 1984 we were predicting that CD would surpass LP in sales in a little over 10 years...about 1995. How long did it really take? Maybe five! I would say that by 2005 two-channel recording will be history. We will be having stereo "revivals" where the excess production capacity of fully amortized plants will have folks squeezing out the last few dollars from a dead technology, as they are now with tubes and LPs.

Daniel Sweeney on Multichannel Formats *(continued from page 72)*

and eliminate matching problems.)

Best Bets

Multichannel music recording is in an early state of development and may well remain in that state for years to come. From my view, the scanty software and insufficient format specifications should indicate extreme caution in making sizable investments in hardware at this time—if music listening is your primary use for your audio system.

If you wish to enjoy video surround-sound tracks, you are better off obtaining a dedicated system than attempting to retrofit your music system. Should that system be THX? I am reluctant to suggest that no uncertified system will do, but I will say that THX-certified systems enjoy the special advantage of being through-engineered in every particular and thus being relatively easy to set up and calibrate. No non-THX system that I am aware of reveals the same attention to every aspect of the signal chain. On the other hand, I am not sure that the THX approach to rear channels is ideal for the new 5.1 format. Unfortunately, no one else is attempting to design for the new format with anything approaching the thoroughness of Lucasfilms, so you can't assume that the THX approach has been automatically obsolesced by the appearance of the new format.

Finally, I would mention again that to obtain the full potential of multichannel playback we need more than five full-range channels. Six would be better, eight would be much better. Extra channels could be obtained by combining matrix technology with 5.1, and John Eargle has in fact suggested that approach. Recently, as I indicated, many of the backers of an audio-only version of DVD have come out in support of an open standard that would permit more than five channels at the discretion of the recording engineer. Nevertheless, the feeling is widespread that the dominating presence of video will dictate the general use of a five-channel configuration for music as well as movies.

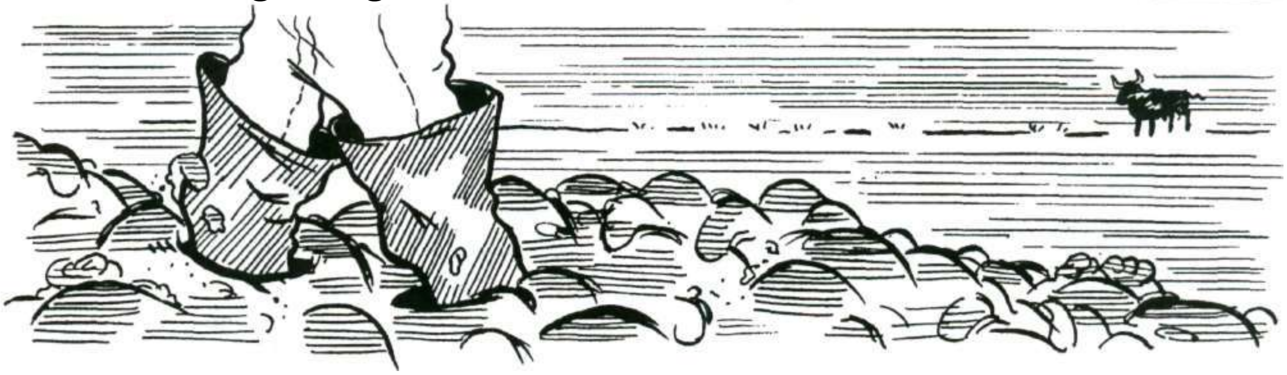
On a practical level, any attempt to introduce multichannel in the perfectionist market must inevitably come up against critical cost constraints. High-quality stereo systems are already inordinately expensive, and very few individuals are likely to treble their expenditures to obtain the extra channels. Hardware manufacturers realize this and many propose to address the problem by lowering speaker quality on the grounds that listeners will be so awed by the extra channels as to ignore nuances of sound reproduction. Indeed, the same attitude is already manifest in the some formats themselves, where extra channels have been obtained at the cost of severe data compression.

I will end by noting that in my many conversations with manufacturers, both high-end and mass-market, the prevailing note in the discussions has been how 5.1 is going to make a killing in the marketplace, while, sadly, few individuals were inclined to mention the specifics of how they were going to exploit the potential of the format to create more convincing musical reproduction. In the midst of these discussions I was frequently reminded of my duty to remain unfailingly enthusiastic about the benefits of the new format—for the good of the industry.

Unfortunately, I remain at some level a consumer and I can't help suspecting that in the frantic effort to cash in, musical ends may be slighted. But in the words of one manufacturer who will go nameless, "You have no right to put your own selfish listening pleasure above the good of the industry." And I guess he's right. •

Hip Boots

Wading through the Mire of Misinformation in the Audio Press



***Editor's Note:** The unprecedented epidemic of self-indulgent subjectivism and untutored expertizing that has infested the audio press (and, on top of it, spawned dozens of worthless new publications) puts this column in a dilemmatic position. What's the point of hacking away at one or two tentacles of the beast when all the others are out there writhing and grabbing? When we started this, the situation was not yet completely out of hand; now it is. I have decided that this time, for a change, we are going to focus on perverse lapses from scientific objectivity by those whom we generally trust—the good guys. If you crave one more fix of bad-guy bashing, see pages 13-15.*

Here is the ultimate example of a good guy going off the deep end and landing in bad-guy territory. Indeed, this is such a heartrending story of a highly regarded academic gone tweak that we can't think of anything to follow it with as an encore, so it will have to stand alone as a solo Hip Boots item. David Rich wanted to do this one; he has the floor:

The Essex Yecch-o *(Et Tu, Malcolme?)*

Only in the ***The Audio Critic*** would you be likely to see a critique of an article that ran in *Stereophile* well over a year ago, in the October 1995 issue to be exact. (Do not look at me but instead please talk to the Editor about our publishing schedule.) In the case of "The Essex Echo 1995" by Professor Malcolm Omar Hawksford, Ph.D., the situation is so strange and bizarre that even this late critique is still entirely relevant. *[I was ready to publish this in Issue No. 23 but David wasn't. Hey, I keep him around for extra circumspection—Ed.]*

The *Stereophile* piece is apparently based on an earlier article with the same title, published in the English magazine *Hi-Fi News & Record Review* in August 1985. (There its subtitle was "Malcolm Hawksford looks at Maxwellian theory & interconnect memories.") Although the article is highly technical, it was never presented at a technical conference and was never published in an engineering or science journal. Now, in the publish-or-perish world of the university that Dr. Hawksford inhabits, this is hard to explain. Here is an article filled with equations that purports to explain why audio cables sound different—oh yes, quite independently of RLC interactions—and its author presents it only through the consumer press! If the conclusions of the article were true, it would be just the thing to bring in grants, best-paper prizes, and bright graduate students. Just the things you need to get to be the head of the department or even the dean of the school.

A close look at the article shows why it never was submitted to a professional journal: it would never stand up to peer review in any such technical journal. Some professional conferences require only that an abstract of a proposed paper be submitted. Under those conditions this work could have been presented. Did Dr. Hawksford not want to stand on the podium and take the questions that would result from the presentation of this work at any professional conference? Why Hawksford should want to be associated with work so full of holes is unexplainable, given that he has done much significant work which has been published in various technical journals and is for the most part highly respected. It looks like he published it in consumer magazines because he knew it could be published nowhere else, but would it not make sense not to publish it at all and not let his reputation be affected by publishing poor-quality work?

Although the intended audience would appear to be the audio consumer reading an audio magazine, the mathematics of this article are at a level that can only be understood by junior-level applied mathematicians, electrical engineers, or physicists. Advanced methods of vector calculus, including curl and divergence, are used. *[That's* curl *with a small* c, *audio freaks; it has nothing to do with John Curl.—Ed.]* Partial differential equations are derived. On top of this, concepts that take weeks to be taught in a junior-level electromagnetics course are presented in a small sidebar. This course is typically considered the most difficult in the whole E.E. course sequence.

Despite John Atkinson's introductory statement that the article is "not as hard to grasp as it looks," I cannot see how a normal reader who has not been exposed to this material could have any idea what is going on. Given the audience, we must guess the intent here is not to teach but to impress—with the knowledge of the author and the complexity of the subject mater.

When the audience is totally uncomprehending of what is going on, as must surely be the case here, it is easy to slip in fiction between the facts without the slightest risk that the reader will feel it going in. Indeed, most professional E.E.s could get fooled. It is interesting to note that much of the argument that is made by Dr. Hawksford was also made David Lindsay of the Lindsay-Geyer cable company. A patent was even awarded to Dr. Lindsay, showing the patent examiner could be fooled by EM sleight of hand.

We can confirm the theoretical analysis is flawed because we cannot find the predicted echoes when we look for them experimentally. We have to go no further than the June 1991 issue of *Stereophile* to find that out. Believe it or not, John Atkinson ran the experiments and had to admit he found no echoes or dispersion effects. Hawksford's article does claim to show experimental results that confirm his theory, but his work is not of the same quality as Atkinson's. (I cannot believe I wrote that last sentence. Wake me up; this must be some kind of bad dream.)

The graphs shown in Hawksford's article lack scaled vertical and horizontal axes. *[Shades of Bruce Brisson's old MIT ads!—Ed.]* Important information on the computer simulation and the experiment is not disclosed. For example, the response characteristic of the transconductor and its g_m are not given. Neither the R, L, and C values of the cable under test nor even its length are disclosed. Independent replication of the experiment is not possible under these conditions. Another important point to note is that Hawksford's cable is terminated into a short, not a resistive load. Atkinson ran his experiment with a normally terminated cable corresponding to real-world use.

Driving the virtual short that the cable represents in the Hawksford experiment requires the transconductor circuit to source and sink a lot of current. High current is required to make the signal levels at the cable's input high enough to be measurable. In Figure 5 of Hawksford's *Stereophile* article the differentiator is shown in great detail but the transconductor is only shown as a block.

I ran a SPICE circuit simulation of Figure 5 and found I could create a plot similar to Figure 7, modeling the cable only as an RL network. The key was to limit the bandwidth of the transconductor to 10 times the tone-burst frequency. Without any details from Hawksford we do not know enough about his circuit, but I bet my estimate is close and he is observing not a novel cable distortion but merely problems in his experimental setup. *[Shades of cold fusion!—Ed.]*

Note that the Hawksford computer simulation is clearly not at steady state before the tone burst is cut off. This is unexpected and makes me think his simulation is flawed, since the effect is not seen in his experimental results nor in my computer simulation

Now then, EM is not my field of expertise, so the Editor and I consulted an expert in the field to give us a better understanding of the sleight of hand used in the analytical part of Hawksford's paper. Our consultant is a tenured professor of considerable distinction, the author of many highly regarded professional papers, who is merely doing us a favor and has no reason to get personally embroiled in a controversy with either the audiophile community or Dr. Hawksford. We were therefore asked to keep the professor's comments anonymous. They are as follows:

> Interestingly, the author's equations, his basic analysis, and the numbers he calculates are correct. It is his interpretations of the meaning of the numbers that are nonsense.
>
> The problems start in the second column on page 55. He refers to Table 2 and the significance of the skin depth and velocity of propagation varying at audio frequencies. This is true. However, the velocity of propagation in the conductor is not relevant, since in electromagnetic theory all of the wave that penetrates the conductor is lost as heat. The velocity of the wave in the conductor is in a direction perpendicular to the surface. This wave is called the Poynting vector and is entirely lost as heat. If the conductor is perfect and lossless, all of the Poynting vector is along the direction of the conductor and none is lost in the conductor, and thus there is no heating. With a finite conductivity, there is a small electrical vector along the direction of the conductor, the I*x*R drop; this results in a small Poynting vector into the conductor, and it is the Poynting vector component in electrodynamic term that causes the wire to get hot.
>
> The heating is not a physical effect as we normally think of it. That is, we normally think of heating in the wire to be due to electrons moving in the conductor and giving up heat of motion to the molecular structure of the wire. But in electrodynamics we do not have to consider such details. We simply use the equations and they give the correct result.
>
> The fact that the velocities of propagation of the axial Poynting vector are tiny has nothing whatever to do with the propagation velocity of the electrical wave moving down the length of the conductor. This velocity is determined only by the properties of the insulating medium which surrounds the conductor and is always a good fraction of the speed

(continued on page 81)

Capsule CD Reviews

By Peter Aczel
Editor and Publisher

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

*David Ranada has been listening to CDs with score in hand, taking **notes,** and planning reviews, but none of it has reached my desk as of press time (despite the unconscionable delay of the latter). He assures me that his reviews are timeless and will be just as good in the next issue. Meanwhile my admittedly less scholarly and probably less durable capsule reviews will have to suffice. Feel free to disagree with me; I don't have a degree in music like David.*

Celestial Harmonies

This Arizona-based company, which is also behind the **Black Sun** label, has all kinds of special-interest and ethnographic projects, such as, for example, "The Music of Armenia" and "The Music of Vietnam," in many volumes, and authentic Australian didjeridoo recordings. I picked just one release for review here because I can relate to it.

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"¡Cante Gitano!" (Nine flamenco selections.) Rafael Jiménez, vocals; Canito, guitar; Paquete, guitar; et al. 13112-2 (1996).

Flamenco, the music of the Andalusian Gypsies, may very well be the most complex and sophisticated European folk music. My flamenco education goes back to Carlos Montoya, Sabicas, Manitas de Plata, that sort of thing. Listening to this very contemporary flamenco release I realized that flamenco has changed as much as jazz and rock. Rafael Jiménez "Falo" is a Gypsy singer now in his early thirties, performing in the flamenco clubs of Madrid. His singing is still basically traditional but shows more non-Spanish influences than I expected (such as Indian tabla accompaniment, Gregorian chant, etc.) He is a very impressive performer, with great vocal equipment and powerful phrasing. The Madrid recording may possibly have been made on analog tape but has great presence and detail. Very quiet background, too.

Delos

If John Eargle were not the engineer behind the new Virtual Reality Recording (VR2) series of Delos, I would be inclined to dismiss the whole thing as a marketing gimmick. After all, there has been no change in microphones and other recording equipment as listed in the CD brochures. My complete trust in John, however, leaves me persuaded that he has indeed started to record subtle cues that provide additional soundstage information, even in plain-vanilla stereo but especially in—pardon the expression—Dolby Surround. The VR2 discs do have a wonderfully spacious, three-dimensional quality without any loss of up-front detail. But then John was making that type of recording for Delos back in 1987! (Well, almost.)

•

***J. S. Bach:** The Six Brandenburg Concertos. The Chamber Music Society of Lincoln Center, David. Shifrin, artistic director. DE 3185 (2 CDs, 1995).*

These are wonderful musicians, none better, but their performance here is a bit lackadaisical, perhaps just because they can play this music faultlessly in their sleep. I like my Brandenburgs played with a little more drive and exhilaration. The VR2 recording is as good as it gets, but the technique makes less difference in chamber music than with large orchestral or choral forces.

*"Double Forte!" Featuring The Alexander Variations by **Calvin Hampton,** plus works by **Saint-Saëns, Wagner, Mozart, and J.C. Bach.** The Organs of the National City Christian Church, Washington, DC; Todd Wilson and David Higgs, organists. DE 3175 (1995).*

A church with two organs is the shtick here, and the beautifully crafted, highly accessible modern work by Calvin Hampton is the centerpiece. This one you should really hear in Dolby Surround to appreciate the interplay of the two organs. The John Eargle recording (not a VR^2) has very nicely delineated organ bass and sounds perfectly natural despite the complex sonorities. I would have been happier with other modern organ pieces, however, not the "Ride of the Valkyries" transcribed for two organs. Please!

•

***Rodion Shchedrin:** The Carmen Ballet. **Georges Bizet:** Carmen Suite No. 1. Orchestre Philharmonique de Monte Carlo, James DePriest, conductor. DE 3208 (1996).*

John Eargle meets Gene Pope; James DePriest meets Mark Gorenstein (see Issue No. 23, pp. 79-80). My vote goes to this new VR^2 release, but not by a wide margin. I think the Delos sound has a bit more immediacy, as well as better bass definition, and the old-line casino capitalists play a bit more securely than the young ex-socialists. The Delos CD also has the advantage of more imaginative programming than the all-Russian one, as the contrast between ur-Bizet and spiced-up Bizet is lots of fun—the pairing is a natural. See also my comments under PopeMusic below.

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***Dmitri Shostakovich:** Symphony No. 8 in C Minor, Op. 65. Dallas Symphony Orchestra, Andrew Litton, conductor. DE 3204 (1996).*

From Dorian to Telarc to Delos—has the excellent Dallas orchestra found a permanent home for its recording affiliation? I hope so because John Eargle has made them sound better in their supposedly state-of-the-art concert hall than anyone else. What's more, Litton himself has said so—in print! This latest VR2 release is better played and at least as well recorded as the earlier Tchaikovsky CD reviewed below. This is one of Litton's favorite pieces of music and it shows. (Under any conductor, No. 8 is a much finer work than the cornball No. 7.) The steamroller third movement in this recording is an unbeatable audio demo.

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***Peter Ilyich Tchaikovsky:** 1812 Overture, Op. 49 (with chorus); The Sleeping Beauty, Op. 66 (9 excerpts, arr. Andrew Litton); The Voyevoda, Op. 78; Moscow (cantata). Dallas Symphony Orchestra & Chorus, Andrew Litton, conductor. DE 3196 (1995).*

This is one of the earlier 20-bit Virtual Reality Recordings and it's good enough to test and/or demonstrate the finest playback equipment. Absolutely stunning, in tonality and in dynamics. The 1812 Overture is played in the arrangement by Igor Buketoff, in which the three Russian vocal pieces quoted instrumentally by the composer are are actually sung by a chorus—hey, why not? The best music here is *The Sleeping Beauty,* played a little foursquare but not enough to hurt it. (Litton has yet to prove to me that he is a conductor of the first rank.) The other pieces are interesting but not great. It's the recording here that's great.

Denon

Is this label (Nippon Columbia) less active in classical recording than it used to be? Somehow, without quantitative information, that's my impression. Whatever they do remains of high quality, in any event.

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***Édouard Lalo:** Symphonie espagnole, Op. 21. **Camille Saint-Saëns:** Violin Concerto No. 3 in B Minor, Op. 61. Chee-Yun, violin; London Philharmonic Orchestra, Jesús López-Cobos, conductor. CO-18017(1996).*

Absolutely breathtaking fiddling. This lovely young Korean-American woman has grown into an artist of major importance. Her tone is her long suit—pure silk, velvet, and polished rosewood—and she hits every note square in the middle, with dazzling facility in the rapid passages and never, never a strained or even momentarily unpleasant sound. I could fault her for the "wow" style—loads of vibrato, portamento, rubato, and every other *-to*—but this is wow-style music, not Johann Sebastian Bach, so all of it is appropriate. Is she as good in Beethoven? I don't know, but this is a hugely enjoyable CD, with outstanding orchestral work and gorgeous sound at all levels from *ppp* to *fff.* (Yes, a "Mastersonic 20-Bit Recording," but good by any other name.)

Deutsche Grammophon

DGG's classical sound used to be awful but now, with 4D Audio Recording, it's really quite good, almost on a level with Delos, Telarc, and the like. No longer hesitate to buy the new CDs of their great artists because of the sound.

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***Achille-Claude Debussy:** Nocturnes; Premiere Rhapsodie pour orchestre avec clarinette principale; Jeux; La Mer. The Cleveland Orchestra, Pierre Boulez, conductor. D108521 (1993).*

Grand slam! Boulez is, in my opinion, the best conductor of French music, certainly the most precise and idiomatic. The Cleveland is all in all the best orchestra on standard repertory, with the most precise ensemble work. Do you think Boulez would bother to make new recordings of played-to-death orchestral staples if he weren't after a world record? You have never heard these pieces played like this. Overwhelming. Even if you own only 20 classical CDs, I suggest that this be one of them.

***Antonio Vivaldi:** The Four Seasons (Spring, Summer, Autumn, Winter). **Fritz Kreisler:** Concerto in C Major in the Style of Vivaldi. Gil Shaham, violin; Orpheus Chamber Orchestra. D 208093 (2 CDs, 1994-95).*

There are many ways to play this much-played music. What we have here is the powerhouse approach—driving rhythm, speed, dynamic contrasts, stupendous precision. The already legendary young Shaham and the incomparable Orpheus both have the "chops" to make such an approach 100% convincing. This is a very exciting performance, the one I prefer to listen to these days. The same artists also made a video of the three "Winter" movements, and there is a bonus CD-ROM (Mac/PC) about the video included with the set. It made my totally *au courant* Power Macintosh crash. I don't care. This is a great recording, with or without a giveaway.

dmp

Tom Jung, the technofactotum of this label, remains the master of ultrahigh-definition, in-your-face audio with small and medium-sized groups. Perfect hi-fi show material.

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"The Robert Hohner Percussion Ensemble World Music Tour." Percussion, lead vocals, & chorus. Robert Hohner, director. CD-514 (1995).

An interesting sequel to the "Different Strokes" CD I favorably reviewed in Issue No. 19. This is a little more ethnic/tribal in program material, featuring West African, Haitian, Afro-Cuban, Trinidadian, and (surprise!) Japanese banging, clanging, and booming, not to mention grunting, mumbling, and chanting. It's more fun than a barrel of monkeys and it's recorded with the expected Jungian impact in Circle Surround, yet another stereo-compatible matrix technology. The only thing I miss here is some very deep (16 to 32 Hz octave) boom-booming, but then I'm an incorrigible subwoofer nut.

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*"DMP Big Band Salutes **Duke Ellington.**" Dave Glasser, alto saxophone; Lew Soloff, trumpet; Britt Woodman, trombone; Dan Rosengard, piano; Lynn Seaton, bass; John Riley, drums; 10 other instrumentalists. CD-520 (1996).*

I inserted this literally hours before we went to press because I was totally blown away by this new release and wanted to be the first reviewer to call your attention to it. I have never heard a big band recorded with such unbelievable fidelity. The instrumental presence and timbral accuracy are simply overwhelming. Most amazing is the quality of the brasses—red-hot and penetrating but without the slightest irritation factor, brilliantly etched and sweet at the same time. *That* I have never heard before. All the other instruments are also detailed to the nth degree and palpably *there.* What a speaker test disc! But that's not all. The 16 musicians are New York's best, playing with breathtaking virtuosity and recreating the Ellington sound and performance style as credibly as I have ever heard—and I am generally not in favor of nostalgic ersatz mimicry. Audiophiles, big-band aficionados, Ellingtonians, drop everything, go out, and buy this CD!

Dorian

This label deals with the flagging interest in classical recordings by releasing a large number of not-quite-mainstream and outright niche-market CDs, most of which involve relatively low overhead. The idea is that a lot of different low-volume sales will add up to a satisfactory bottom line. You and I are the beneficiaries of their highly varied—shall we say Greek-diner style?—menu.

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*"An American Tapestry." Orchestral works by **William Schuman, Griffes, Ives, Hovhannes,** and **Piston.** Dallas Sympnony Orchestra, Andrew Litton, conductor. DOR-90224 (1995).*

Dorian's last hurrah in Dallas, and maybe Litton was right—this is probably not the sound of the orchestra from the best seat in the hall. The recording is far from bad, needless to say, but there is a certain lack of immediacy, as if you were in a good cheap seat, up there somewhere. Aside from that, Litton does not have quite the same feeling for these robust, no-nonsense American works as Gerard Schwarz, for example. This gets a grade of B minus from me.

•

***Edvard Grieg:** The Three Violin Sonatas. (No. 1 in F Major, Op. 8; No. 2 in G Major, Op. 13; No. 3 in C Minor, Op. 45.) Ivan Zenaty, violin; Antonin Kubalek, piano. DOR-90234 (1995).*

These lovely violin sonatas used to be frequently played but today they are neglected, perhaps because of their unsophisticated Romanticism. If all performers had played them the way these two Czechs do, they might have remained popular. I have long ago formed the opinion that in nineteenth-century Romantic music one Czech is worth seven Austrians, and this recording confirms that. Irresistibly charming and stylish playing. What's more, the violin is so accurately reproduced on this CD that you can use it as a tweeter and midrange test. What more can you ask for?

*"The Italian Lute Song," featuring works by **Monteverdi, Frescobaldi, Caccini, Carissimi,** et al. Julianne Baird, soprano; Ronn McFarlane, lute. DOR-90236 (1995-96).*

This one is easy to like. Julianne Baird would sound utterly enchanting just singing "Three Blind Mice," and nobody plays the lute more musically and authoritatively than Ronn McFarlane. The music here ranges from pretty to superlative, whereas the performances are uniformly wonderful. Baird's voice remains a miracle, at least in this tonal and dynamic range, and McFarlane is likewise orphic. The recording is absolutely crystalline, one of the finest Dory/Troy efforts, demo quality and then some. I promise you this: when this CD is over, after 70 minutes, you will not have had enough.

"Lullaby Journey," Custer LaRue, soprano; Chris Norman, wooden flute; Kim Robertson, Celtic harp. DOR-90232 (1995).

I discovered this collection of traditional lullabies at the CES, where the CD was used by an exhibitor as a resolution test. On the opening track, LaRue vocalizes *ppp* under the pungent tones of Norman's wooden flute. You can just barely hear it and only on the best systems. The unique timbre of the wooden flute on the various tracks is an audio feast by itself. On top of it, these are excellent musicians; LaRue has a very sweet voice; and the music is invariably charming. It is almost superfluous to add that the Dory/Troy recording is state-of-the-art.

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***Heitor Villa-Lobos:** Symphony No. 4 ("Victoria"); Cello Concerto No. 2; Amazonas. Simón Bolívar Symphony Orchestra of Venezuela, Enrique Arturo Diemecke, conductor; Andrés Díaz, cello. DOR-90228(1995).*

If you only know the *Bachianas brasileiras* of Villa-Lobos, check out this *huge* symphony (1919) for *huge* orchestra. Interesting, at the very least, and occasionally powerful. The concerto (1953) is finely crafted and very accessible. Díaz is an excellent cellist. The recording is very good, but Dorian can do even better (maybe not in Caracas). The least important, highly derivative *Amazonas* ballet music is actually the best recorded.

Erato

Atlantic Records, distributors of this label (and of Teldec and Finlandia), are now restricted, I've been told, to 13 review copies of each new release for the entire USA, as an economy measure. I am clearly not included in that apostolic number. (I think they save approximately the price of a Big Mac with each withheld CD.) The release reviewed here is left over from the prodigal old days.

•

***J. S. Bach:** Mass in B Minor, BWV 232. The Amsterdam Baroque Orchestra & choir, with soloists; Ton Koopman, conductor. 4509-98478-2 (2 CDs, 1994).*

I am attuned to the Bach performance style of my neighbors, the superb Bach Choir of Bethlehem (Pennsylvania, that is), so that the pinched period-practice rendition of the B Minor by the Amsterdammers leaves me somewhat cold. The ensemble sonorities are quite beautiful, the textures are crystal clear, the choir sings well, but the soloists are mediocre (the countertenor is a disaster), and the overall impact leaves a lot to be desired. The recording, on the other hand, is wonderfully transparent.

Harmonia Mundi

The following CD was sent to me (and presumably to other reviewers) completely unsolicited, a most unusual gesture by this label. Their usual policy is to send a "digest" disc with sample snippets. Obviously they expect Part to play an important part in their sales. (Sorry!)

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***Arvo Pärt:** De Profundis (Psalm 129); Missa Sillabica; Solfeggio; "And one of the Pharisees"; Cantate Domino (Psalm 95); Summa (Credo); Seven Magnificat Antiphons; The Beatitudes; Magnificat. Theatre of Voices, Paul Hillier, artistic director. 907182 (1996).*

If I'm going to buy into the current chant craze, I might as well buy into Arvo Pärt, whose neo-medieval/postmodernist chants are more interesting than most. This is the kind of music that either fascinates you or turns you off totally. These works cover the period from 1964 to 1990 (Pärt was born in 1935) and are beautifully sung, Hillier is a Pärt specialist, and the composer was there at the recording sessions. The sound is superbly transparent, as it usually is when Robina Young is the producer. Good test CD for massed voices.

London

A London (i.e., English Decca) recording is always a known quantity audiowise; we know it will be nonminimalist, with lots of microphones, as well as a bit on the bright side, but almost surely the best of *that* kind, detailed and euphonious.

***Richard Wagner:** "Orchestral Favourites." Prelude to Die Meistersinger; Ride of the Valkyries; Dawn, Siegfried's Rhine Journey, & Siegried's Funeral Music from Götterdämmerung; Tannhäuser Overture and Bacchanal; Lohengrin Act III Prelude. Royal Concertgebouw Orchestra, Riccardo Chailly, conductor. 448 155-2 (1995).*

The last thing the music world needs is another recording of these played-to-death orchestral standards—except when the performances are this good. Wow! Why does it take an Italian conductor to play Wagner with this kind of rectitude, clarity, and rhythmic sophistication? Shades of the Old Man! The *Meistersinger* prelude is paced and phrased exactly right for a change, and the *Tannhäuser* Bacchanal (Paris version) is nothing short of dazzling. I could praise the other pieces equally. Of course, it helps to have a great orchestra. The brasses of the Concertgebouw are weightily warm and brilliant at the same time, a unique sound. The strings are marvelous. On top of everything else, the recording in the Grotezaal has rounder edges than is usual for Decca and John Dunkerley; that really puts the icing on the cake. Wagnerians, add this duplication to your collection.

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***Richard Wagner:** Das Rheingold. Robert Hale, Wotan; Hanna Schwarz, Fricka; Kim Begley, Loge;*

Peter Schreier, Mime; Franz-Josef Kapellmann, Alberich; et al. The Cleveland Orchestra, Christoph von Dohnányi, conductor. 443 690-2 (2 CDs, 1993).

Yet another complete *Ring* in the offing, but I'll wait for *Die Walküre* before I develop too strong an opinion about this one because *Das Rheingold* is basically just an orchestral piece (a long, long one) with some singing actors thrown in. No heldentenor, no bigtime fat lady. We'll have to see whom they pick for the rest of the cycle. As far as the orchestral performance goes, the Cleveland is America's best, so naturally they play magnificently. Dohnányi's pacing is excellent; the recording in Severance Hall is first-rate Decca; thus there are no serious demerits so far. The competition is fierce, however.

Mapleshade

I must confess that I don't have much affinity for the particular type of jazz and blues cultivated by this label—that's my problem, not theirs—but I stand in awe of Pierre Sprey as a recording engineer. I don't even share his proanalog ideology—he starts with a live-to-2-track analog master tape—but I can't argue with the sound he obtains. I have never heard more startlingly lifelike recordings of small groups playing this kind of music. Perfect audio testing and demo material—pick any title at random from their catalog.

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"Sweetman with his South Side Groove Kings: Austin Backalley Blue." Sweetman, tenor sax, and his band. Wildehild MS 02752 (1992).
"Bad Influence." Live at the Bad Habits Cafe. Whop Frazier, vocals & electric bass; Junior Tash, electric guitar; et al. Wildchild MS 03152 (1993).

Wildchild is Mapleshade's alternative label, subtitled "Raw Music, No Additives." That says it all. The Sweetman disc is a studio recording of a truly raunchy-sounding Texas blues band, with an incredibly sleazy '50s stripjoint style tenor sax in the lead. The Bad Influence disc is an on-location recording of an equally slummy type of blues/rock band, playing at a beer-and-ribs joint in the Washington, DC area. I have no idea how connoisseurs of this sort of music would rate these groups against the competition; for me they are a perfect escape into surrogate squalor and great junk food for my reference stereo system.

Marco Polo

This label is now entirely under the control of, and virtually indistinguishable from, Naxos, so all my applicable general comments will be found under that heading.

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***László Lajtha:** Orchestral Works. Vol. 1: Symphony No. 7, Op. 63; Suite No. 3, Op. 56; Hortobágy, Op. 21. Vol. 2: Capriccio (Suite de ballet), Op. 39. Pécs Symphony Orchestra, Nicolás Pasquet, conductor. 8.223667 and 8.223668 (1994).*

It's a ha-ha in a *Seinfeld* episode that the characters can't recall the name of "the third tenor." Quick, who was the third Hungarian composer of the 20th century? Lajtha (pronounced approximately like *loiter* without the *r*), that's who. Bartók was an immortal genius; Kodály was a master; Lajtha (1892-1963) was a superb craftsman, as this no longer very new pair of CDs shows. Almost any music lover will hugely enjoy the pungent rhythms and colorful orchestration of this music. It isn't boring. Great originality? Maybe not, although the 7th Symphony has moments of genuine eloquence. The provincial orchestra is amazingly competent here, and the Hungarian recording is on the Telarc level.

Melodiya

BMG Classics (owners of RCA Victor) continue to distribute their high-tech restorations of important performances from this great Russian catalog. (See also Issue No. 23.)

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*MRAV1NSKY EDITION, Volumes 11-20. Featuring: **Beethoven, Brahms, Bruckner, R. Strauss, Shostakovich, Ovsyaniko-Kulikovsky, Tchaikovsky, Glazunov, Rimsky-Korsakov.** Yevgeny Mravinsky, with the Leningrad Philharmonic Orchestra. 74321 29459 2 (10 CDs, 1947-73).*

The second set of Mravinsky reissues dates much further back than the first, so you can expect less good but still quite acceptable recording in the 1947 Shostakovich Eighth *(the* authoritative reading) than in the completely up-to-date-sounding Brahms Fourth of 1973 (glorious performance). The great conductor and his great orchestra continue, posthumously, to astonish; something vital is happening in every bar of every piece here. Definitely hall-of-fame stuff, not to mention a triumph of remastering.

MusicMasters Classics

One thing I like about this label (another province of the BMG Music empire) is that they have no fluff in their catalog, at least none that I'm aware of. They seem to record only important music, as performed by important artists. Can you make a living that way?

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***Ludwig van Beethoven:** Sonatas for Violin & Piano. Vol. I: A Major, Op. 47 ("Kreutzer"); D Major, Op. 12, No. 1. Vol. 11: A Major, Op. 30, No. 1; C Minor, Op. 30, No. 2; G Major, Op. 30, No. 3. Claude Frank, violin; Pamela Frank, piano. 01612-67087-2 (1992) and 01612-67106-2 (1995).*

The Franks are a father-and-daughter team of great distinction. Their playing is unfailingly musical, lovely in sound, technically secure and then some, but not without occasional fussy mannerisms. There is so much competition in this music that I can't possibly rate these obviously excellent performances against the rest of the field. Max Wilcox was the producer and engineer, and that means utterly natural audio, with a high degree of resemblance to live sound.

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***W. A. Mozart:** String Quartets, Volumes I to III. The American String Quartet: Peter Winograd & Laurie Carney, violins; Daniel Avshalomov, viola; David Geber, cello. 01612-671xx-2 (3 CDs, 1992-95).*

There may exist more brilliant or scholarly traversal of the Mozart string quartet repertoire but none better-sounding, I'm sure. These four excellent musicians play on a matched set of Antonio Stradivari instruments, on loan from the Smithsonian Institute especially for these recordings. The resulting string tone is incredibly beautiful, always sweet on top and darkly warm in the lower ranges. The three volumes issued so far cover nine quartets, with a balanced program of one early, one middle-period, and one late work on each CD. Three of the famous six dedicated to Haydn are here (K. 387, K. 421, and K. 428), as well as the superb K. 499 from the same period. The performances are uniformly lucid, idiomatic, and unexaggerated. Judy Sherman produced and engineered the recordings, with obvious competence as manifested by the gorgeous Stradivari tone.

***Igor Stravinsky:** The Composer, Volumes I to VIII. The Orchestra of St. Luke's, Robert Craft, conductor, with soloists, choral groups, etc. 01612-67xxx-2 (10 CDs, 1991-95).*

Robert Craft, as everybody knows (well, everybody I associate with), was Igor Stravinsky's musical Boswell, from 1948 until the composer's death in 1971. They were practically inseparable. A Robert Craft performance of a Stravinsky composition is the nearest thing to a Stravinsky performance. That doesn't mean Craft is a great conductor—nor was Stravinsky—but we are very lucky to have these unchallengeably authoritative interpretations in 1990s sound. St. Luke's is a virtuoso orchestra of New York freelancers and moonlighters, and Craft doesn't have to be a Fritz Reiner or Pierre Boulez to make them play with spirit and precision. Eight volumes on ten CDs cover a lot of ground, but the series is far from complete; indeed, more than a few major works are still missing, such as *Petrouchka* and *Le baiser de la fée.* We have more to look forward to. (Speaking of Reiner, his incomparable and Stravinsky-approved 1958 recording of the Divertimento from *Le baiser* may be one reason for Craft's postponement of a new recording.) All eight volumes issued so far were produced and engineered by Gregory K. Squires. The sound is clean, dry, and precise throughout (just like the music and the conducting). I have no problem with that; I actually prefer the "analytical" kind of recording when it suits the program material, and this is a fine example of the genre. Bottom line: the Stravinsky devotee obviously has other options in individual opuses here and there but none in a "complete works" set. This is the one. And that's far from regrettable.

Naxos

While the classical CD business founders, scrimps, and kvetches, Naxos thrives, bigtime. There must be a reason, right? The Naxos formula is, to paraphrase the advertising slogan of Mamma Leone's New York restaurant: "Record good classical music and give people plenty of price incentives. They'll come." A student who can't afford the DGG recording of a favorite symphony will buy the Naxos version at a fraction of the price, and it is likely to be a very respectable performance, in state-of-the-art sound, quite possibly better recorded than the DGG. I recently visited the Phoenix Studio in Budapest, one of the busiest Naxos contractors. The equipment used by the Hungarians is as up-to-date as anything I've seen anywhere. They just charge less than the Western studios.

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***Ludwig van Beethoven:** Symphony No. 1 in C Major, Op. 21; Symphony No. 6 in F Major, Op. 68 ("Pastoral"). Nicolaus Esterházy Sinfonia, Béla Drahos, conductor. 8.553474 (1995).*
Symphony No. 3 in E-flat Major, Op. 55 ("Eroica"); Symphony No. 8 in F Major, Op. 93. Nicolaus Esterházy Sinfonia, Béla Drahos, conductor. 8.553475 (1995).

It is an essential part of the Naxos credo not to record and release any piece of music more than once, but here they made an exception. Their original Beethoven symphony series (conducted by Michael Halász and Richard Edlinger) wasn't really competitive with all the world-class performances available on full-priced labels. These new recordings definitely are. Drahos, who was a highly regarded flutist before he became a conductor, is a truly superior musician. I endorse his readings of No. 1, No. 6, and No. 8 without any reservations. Every phrase is lovingly inflected and in perfect balance with every other phrase. Every bar dances and sparkles, maybe in a somewhat Haydnesque way (Drahos has recorded a lot of Haydn) but very appealingly in any event. The "Eroica" is another story. It sounds like Haydn's greatest symphony here; the titanic rhetoric is minimized. Maybe that's authentic 1803 style, but I'm used to the monumental interpretations of the great modern conductors. The miniaturized impression is partly due to the meager string section of the orchestra, but not entirely. Drahos wants it that way. His woodwinds, on the other hand, are nothing short of world-class. The best wind players in the country are moonlighting here for their former colleague. The recorded sound is in every way exemplary; I used it to demonstrate to a friend the timbral accuracy and imaging of an outstanding speaker system. It would seem that the Phoenix Studio recording team has solved the far from inconsiderable difficulties of the Budapest hall where the recording was made (called the Italian Institute).

PopeMusic

Gene Pope III is the ideal recording guru for audio tweaks. Like them, he is a True Believer. He believes in a single pair of B & K omnis and Cello electronics, and that's that. If it works—and in many cases it works beautifully—he is a hero. If it doesn't work—and in certain cases it can't—he'll never admit it. His audio ideology and rhetoric will always be strutting, and sometimes stumbling, one step ahead of the pragmatic realities of the concert hall. Give me an engineer who doesn't believe in anything except the end result...

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Peter llyich Tchaikovsky: *Symphony No. 6 in B Minor, Op. 74 ("Pathétique); Francesca da Rimini (Symphonic Fantasy), Op. 32. Russian Symphony Orchestra, Mark Gorenstein, conductor. PM 1006-2 (1995).*

Here is a case in point. It is extremely difficult to get such richly orchestrated music exactly right with just two microphones. The recording sounds thin and shallow, with wimpy bass, in many passages but, strangely, not all. It's almost as if they had experimented with the microphone placement in the middle of the symphony—is it possible? Or could it be level related? The *Francesca* is a 20-bit recording versus 18-bit in the symphony—it doesn't seem to make too much of a difference. Gorenstein is poetic and highly eccentric by turns, almost as baffling as the engineering. The young orchestra plays quite well but lacks the refinement of the top-notchers. I'd give the total effort a squeak-by B minus.

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Ludwig van Beethoven: *Concerto for Piano and Orchestra No. 5 in E-flat Major, Op. 73 ("Emperor"); Piano Sonata No. 14 in C-sharp Minor, Op. 27, No. 2 ("Moonlight"). Naum Starkman, piano; Russian Symphony Orchestra, Mark Gorenstein, conductor. PM 1004-2 (1995).*

I deliberately departed from the alphabetical order of composers here to be able to point out that this is different from the above. The minimalist recording technique works very well with Beethoven's simpler, more nearly monochrome orchestration. The strings have proper weight and the piano is nicely focused. What's more, the veteran Starkman is a very fine pianist who should be better known in the West. These performances are competitive with just about any. If you want to try a Gene Pope production, I recommend this one.

RCA Victor Red Seal

The sound engineering has been getting better and better on this venerable (and now BMG-Music-owned) label. Until recently one could rely on them for OK but not great audio, but slowly the audiophile labels competing in the same market are beginning to have reason to be worried.

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Aaron Copland: *Symphony for Organ and Orchestra (1924); Dance Symphony (1929); Short Symphony (1932-33); Orchestral Variations (1957). Saint Louis Symphony Orchestra, Leonard Slatkin, conductor. 09026-68292-2 (1993-95).*

This is terrific, nonboring 20th-century music, played by an orchestra that deserves to be ranked with the Big Five to make it the Big Six—but the CD's current reputation is based on the engineering. I have to dissent, albeit only mildly. The recording does have tremendous presence and lower midrange impact, as well as superb bass, but the treble is edgy when it gets loud. That's a very common failing of the big commercial labels, probably the result of trying to make everything sound good on every system, including car stereo. Other than that the sound is indeed outstanding, but I favor the recording because of the unique artistic quality of early Copland and the virtuoso playing.

Richard Wagner: *Lohengrin. Ben Heppner, Lohengrin; Sharon Sweet, Elsa von Brabant; Eva Marton, Ortrud; Sergei Leiferkus, Friedrich von Telramund; Jan-Hendrik Rootering, The King; Bryn Terfel, Herald. Bavarian Radio Symphony Orchestra and Chorus, Sir Colin Davis, conductor. 09026-62646-2 (1994).*

What if Wagner had stopped with *Lohengrin* and had never composed the *Ring, Tristan, Meistersinger,* and *Parsifal?* How great a composer would we consider him to be? Maybe as great as Weber? Or Meyerbeer? Or Gounod? Think about it. This is basically a rip-roaring grand opera requiring some good singers. The Grail music (in the prelude and elsewhere) foreshadows the Wagner to come, and the orchestral mastery throughout is a given, but what an incredible leap the *Ring* was! I like the plot, though; intervention by a masked stranger (isn't he?) to help the innocent in distress always works for me. This is a very good performance indeed. Ben Heppner is arguably the best tenor around; his voice does not possess the steel of a true heldentenor but it always sounds beautiful, and that's a lot better than some shrill mediocrity shouting his head off. Everybody else is also good, except Marton (but that isn't sufficient cause to turn me off as it did some critics). Davis is a highly sophisticated conductor, and the Bavarian orchestra is excellent. What more do you want? Even the recording is very close to the best work being done in opera today. I have no problem recommending this set.

Telarc

Yes, there is definitely such a thing as "the Telarc sound," and not everybody likes it as much as I do. It puts texture before structure (tonality before imaging would be another way of saying it), and it tends to be on the dry side. Those happen to be my preferences; I have a bit of a problem with the typical audiophile partiality to the "wet" sound. Two-channel reproduction loses too much information in all that "wetness." (Five-channel/discrete will probably be another story.)

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Joseph Haydn: *Quartet in D Major, Op. 76, No. 5.* ***John Corigliano:*** *String Quartet (1995). Cleveland Quartet: William Preucil & Peter Salaff, violins; James Dunham, viola; Paul Katz, cello. CD-80415 (1995).*

It seems this is everybody's favorite chamber-music CD among the more recent releases. The Haydn quartet is one of those astonishing later works that presage Beethoven; the Corigliano is a stunning post-Bartókian composition, full of arresting string effects and recorded for the first time here. The Cleveland Quartet disbanded (amicably) in 1996 to pursue individual careers; this is titled "The Farewell Recording." I must say they play as if their lives depended on it—magnificent performances. Their Beethoven quartet series (which I also recommend as one of the best) will soon be completed with new releases recorded before the breakup. Judy Sherman was the producer of all the Cleveland recordings and Jack Renner the engineer. This last effort is probably their best; the sound is truly beautiful.

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Zoltán Kodály: *Háry János Suite; Dances of Galánta; Variations on a Hungarian Folksong (The Peacock). Atlanta Symphony Orchestra, Yoel Levi, conductor. CD-80413 (1995-96).*

You should check out this CD mainly on account of the *Peacock* variations, a near masterpiece in what is probably its best digitally recorded version. The *Háry* suite has had more amusing interpretations. The orchestra plays beautifully, and the audio is a 20-bit Michael Bishop job, which is hard to beat if you like the Telarc sound.(I do.)

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Edward MacDowell: *Piano Concerto No. 2 in D Minor, Op. 23.* ***Franz Liszt:*** *Piano Concerto No. 1 in E-flat Major; Piano Concerto No. 2 in A Major. Andre Watts, piano; Dallas Symphony Orchestra, Andrew Litton, conductor. CD-80429 (1995).*

We all know that André Watts can play these Romantic piano concertos in his sleep, with virtuosity and panache. That's a given, but here he "kicks it up a notch" (as Emeril would say), and the results are exciting. What is not a given is that the recording is by Telarc. They ended up being a halfway house for the Dallas orchestra between Dorian and Delos. I think the reason must have been Litton, not the work of Michael Bishop, because the sound is wonderful. Too much "generic" Telarc and not enough hall specifics? Maybe. Personally, I like my CDs a little dry and don't want more hall sound than this, but the Dallas people are pretty chauvinistic about their McDermott Hall (at the Meyerson Symphony Center) and probably disagree with me.

Gustav Mahler: *Symphony No. 9. Cincinnati Symphony Orchestra, Jesus López-Cobos, conductor. CD-80426-A/B (2 CDs, 1996).*

I listen to Mahler with a great deal of pleasure but I am not one of those who believe he was a colossus. His colossal effects come with too much panting and sweating for my taste, especially in the later symphonies. That doesn't mean I don't admire his craftsmanship, melodic invention, and orchestration. Since I am not willing to go to the barricades about Mahler issues, I refuse to start a controversy regarding the relative merits of this new recording in comparison with famous Ninths by Karajan, Bernstein, Walter, etc. I think López-Cobos does a great job clarifying this difficult work, and his orchestra plays very well indeed. The Spatializer-enhanced 20-bit recording is credited to Erica Brenner as producer and Thomas Knab as recording engineer, an apparent changing of the guard at Telarc. The sound is superb, regardless; I can't remember another Mahler recording with this kind of presence and bass.

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W. A. Mozart: *Don Giovanni, K. 527. Bo Skov-*

hus, Don Giovanni; Alessandro Corbelli, Leporello; Christine Brewer, Donna Anna; Jerry Hadley, Don Ottavio; Felicity Lott, Donna Elvira; Nuccia Focile, Zerlina; Umberto Chiummo, II Commendatore/Masetto. Scottish Chamber Orchestra & Chorus, Sir Charles Mackerras, conductor. CD-80420 (3 CDs, 1995).

This is the fourth in Mackerras's monumental series of Mozart operas, recorded within a four-year period with basically the same very consistent and reliable group of performers and equaling in many respects the immortal Glyndeboume 78-rpm recordings of the mid-1930s. My comments in the last issue about the *Figaro* performance apply here equally—Mackerras has no truly great voices at his disposal, but the singing always remains on a high level of technical competence and musicianship. Thanks to the conductor's masterful grasp of the score, the music is imbued with unflagging vitality and polished grace from bar to bar and phrase to phrase. The style is halfway between period practice and the grand tradition, and that suits me just fine. Vocal lines are highly ornamented, and fortepiano (rather than harpsichord) continuo is used. In the title role, Bo Skovhus is quite suave and authoritative but still no Pinza or Siepi. The best voice in the cast is probably the fresh, young, secure soprano of Christine Brewer as Donna Anna; the usually excellent Jerry Hadley, on the other hand, is a little coarse as Don Ottavio. The Jack Renner sound is a known quantity from the previous three recordings in the series: it is close to perfection. Even if this is not the greatest *Don Giovanni* ever, as a "matched" set of four Mozart operas in the same interpretive mold the Telarc series is unbeatable.

"Bop! Frank Morgan with the Rodney Kendrick Trio." Frank Morgan, alto saxophone; Rodney Kendrick, piano; Curtis Lundy, bass; Ray Drummond, bass (last cut only); Leroy Williams, drums. CD-83413 (1996).

This capsule review was initiated by subscriber Robert Burko of Milwaukee, WI, who thought that my affinity for 1940s and 1950s jazz might favorably dispose me toward this new CD. Well, Frank Morgan is probably the last giant left over from the classic bop days and he is still a joy to listen to, as good as ever. The support he gets here, however, isn't what he deserves. Bop is supposed to be driven by the rhythm section, and they are a bit tame (lame?) here. The piano is likewise so-so. Only in the last cut, Thelonious Monk's famous "52nd Street Theme," does the pulse pick up, thanks to the cameo appearance of the bassist Ray Drummond (see also Issue No. 16, p. 68). Morgan's gorgeous tone and superbly authentic bop style are worth the price of admission, in any event. Also, Jack Renner's recording is as natural and transparent as you'll ever hear.

Teldec

See my comments under Erato regarding the difficulty of getting CDs for review from Atlantic Records. The one below came in before the rationing went into effect.

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Franz Liszt: *Mazeppa; Mephisto Waltz.* ***Zoltán Kodály:*** *Theatre Overture; Háry János Suite. New York Philharmonic, Kurt Masur, conductor. 9031-77547-2 (1992-95).*

What we have here proves a number of things. One is that the New York Philharmonic is once again a great orchestra under Kurt Masur. Another is that producer Martin Fouqué has an uncanny knack for obtaining audiophile-quality recorded sound in that utterly dreadful Avery Fisher Hall. Yet another is that the best part of the *Háry János* music, now called "Theatre Overture," is hardly ever played. The CD is worth getting just for that. Is the Suite itself better performed here than by Levi/Atlanta on Telarc (see above)? Well, yes; it's an even better orchestra, for one thing; but Masur is a bit too serious (Kodály wasn't). The Liszt pieces have splendid sweep, their vulgarity redeemed by the great playing. 0

Hip Boots *(continued from page 76)*

of light (typically 80% or more). Thus any dispersion is negligible, as has been pointed out by others many times in the past.

In the same way, the skin depth is irrelevant since it is only a measure of the depth to which the Poynting vector penetrates the conductor. That is, the heat is generated in the region of penetration. However, all of the energy still turns into heat and is lost forever.

By these same concepts, all of the electromagnetic wave (energy or information) that reaches the far end of the wire travels in the space outside of the conductor. This may seem a strange concept, but when using Maxwell's equations, one must obey all the rules and also interpret the results using electrodynamic concepts. Grungy stuff like electrons bumping about in the conductor are not needed and are not part of electromagnetic theory. Isn't this interesting? I repeat. All of the energy travels by means of the electrodynamic fields in the space outside of the conductor! The conductor simply guides the energy. Now you understand why students hate electrodynamics and go into circuits and computers instead.

Similarly, anything that takes place inside the conductor such as boundaries, grains, and all that junk are irrelevant, since any energy that enters the surface of the conductor is lost as heat anyway. It does not matter how it is lost, since it never again contributes to the information traveling along the axis of the conductor.

As for the author's conclusions, they are for the most part nonsense based on a serious misunderstanding of electromagnetic theory and incorrect interpretation of its results. I find it hard to believe this was written by a Ph.D. who is a tenured professor, since it is such garbage.

What is particularly interesting is that this is not the first time John Atkinson will be hearing the above critique. Dick Olsher said basically similar things in a followup to his Lindsay-Geyer cable review *(Stereophile,* August 1991). Of course, Olsher continued to believe the cables sounded as good as he had thought they did when he still believed the incorrect analysis given to him by the manufacturer. Some people just will not give up their belief system but will keep insisting that cables sound different even when theory and experimental evidence crash right before their eyes. Perhaps this is the same belief system that drives Dr. Hawksford to want to present work he knows would never make it into a professional journal.

At a time when 6 Mbits of data can be sent down normal phone lines using the new ADSL systems, any unknown phenomena that distorted signals through wires would have been identified and reported in technical journals. (The speed limit for normal modems is not the wire but the data converter used in the central-office line cards that sample 8 bits at 8 kHz. In ISDN and ADSL systems the speech line card is replaced with a line card designed to receive high-speed data.) These data communication systems would not work if the phenomena discussed in Dr. Hawksford's paper existed.

Once again we see that the high-end audio cable industry is a giant fraud that depends on the fallibility of hearing in open-loop listening tests, frequency response changes due to RLC effects, and finally pseudoscience such as presented in "The Essex Echo."

—David Rich

Book Reviews

By the Staff of *The Audio Critic*

The Complete Guide to High-End Audio
By Robert Harley
Self-published, 450 pages, $29.95

Reviewed by David Rich

The writings of Robert Harley in *Stereophile* come under scrutiny in almost every issue of ***The Audio Critic.*** Much of the misunderstandings and misconceptions in high-end audio can be traced to the pen of Mr. Harley. Since *Stereophile* has so many more readers than this magazine and comes out regularly every month, Harley has a large and powerful pulpit to spread his pseudoscience. I feel it is important for ***The Audio Critic*** to point out Harley's mistakes, lest they become a permanent part of audio lore. Look at the mess he has made of the subject of jitter.

Now Bob Harley has written a book called *The Complete Guide to High-End Audio.* The book is self-published; Bob's wife Evalee runs the "publishing company." She was smart enough not to give me a copy for review but she did sell me one. Bob even signed my copy for me, so he cannot blame his wife for handing me the knife that I will use to lay him bare. Evalee tried to make the point that the book is good for the industry, so I should not review it negatively, and that is clearly true right down to the subchapter "Your Relationship with the Retailer." This is a book that has as its goal the promotion of the high-end industry, not aiding the consumer to get the best-performing products at the lowest cost. What Mrs. Harley does not understand is that I am not part of the industry. I am a consumer advocate and I have no interest in seeing this book do well so it can act as some sort of growth enabler for the high-end industry.

The book has been heavily promoted in *Stereophile,* is blurbed by members of the *Stereophile* staff, and appears to be a success. A page-by-page analysis of the errors in this book would consume this whole issue of our journal. Anybody who purchases the *Guide* gets what is coming to him. It is filled with the audiophile nonsense that we expect from subjective audio reviewers: "Many cables overlay the treble with a coarse texture," etc., etc., etc. How could anybody believe in this stuff? Well, *The Complete Guide to High-End Audio* actually shows you how somebody can believe in this stuff. It has nothing to do with golden ears and everything to do with lack of knowledge of the subject one is engaged in.

This book clearly shows that Harley does not command even elementary concepts in electrical engineering—or in pedagogy. Let's start in the preamp chapter. Figure 5-6 is his block diagram of a preamp. Missing are the input selector, the tape monitor loop, the volume and balance controls, and all other major functional blocks like filters and tone controls. Instead what is shown is a block diagram of an amplifier, not a preamp. .Harley does not understand the difference. But wait, his amplifier (a.k.a. preamp) has no basis in reality. We are told the input stage has no gain; it only does impedance transformations. We are told that FETs may be used in this stage for low noise. Just on the basis of these two explanations we realize that Harley has no knowledge of basic amplifier design or device physics.

Figure 5-7 is a schematic of a line-level amplifier. Harley chooses the truly weird and complex circuit from Audio Research. The input stage has significant gain, but Bob is happy to identify the second stage as the one with all the gain. He then states the next three devices in the schematic are a constant current source. All wet! Two of the transistors are protection devices for the third, which is the output source follower. He identifies the next transistor as the output stage, but that one is the current source! He states that a single blocking capacitor (C40) prevents dc at the output, but the schematic shows two paralleled capacitors C40 and C42. It is clear that he has no idea what is in this schematic. He took notes form someone else and then got it all wrong. The errors Bob Harley is making are very, very basic. A high-school student with an interest in electronics could tell him he got it wrong.

As I said, the circuit Harley uses is about as weird as they come. Why does Harley choose it? It makes it impossible for the reader to understand what is going on. One would want to choose a simple circuit if one were trying to explain to someone new to electronics how a line stage works. In the circuit Harley uses, a unity-gain buffer is in the feedback loop (do not look at me, this is an ARC design). The buffer is formed by an op-amp and a 6-transistor FET output stage. Normally the feedback loop is made with two resistors and maybe a compensation capacitor. In the ARC design the feedback loop is more complex than the amplifier itself. Harley spends two lines on what feedback is but makes no attempt to explain how it works (does he know?). He leaves the reader with the impression that a complex circuit is required for implementing feedback and then moves on to explain that a differential amplifier converts balanced signals to unbalanced signals—wrong again. Trust me, this is what he thinks, as Figure 5-9 shows an XLR input connected directly to a block he marks "Differential Amplifier." It must be Harley's goal to make his reader as confused as possible, or else the reader will question statements like "At the very highest level of music reproduction, there's not even a debate: LP is musically superior to CD."

To make sure the reader remains confused, there follows the schematic of the power supply for the preamp. Again, a weird ARC design which has very high voltage drops on the pass transistors is shown. We are told "The power supply, shown in Fig. 5-8, is quite elaborate—in sharp contrast to the audio circuit's apparent simplicity." Much of the complexity in the circuit shown is in the time-delayed muting circuit not discussed at all by Harley. He then moves on to discuss fully balanced preamplifiers. Figure 5-10 shows two completely separate signal paths for the positive and negative signals. It is clear, after one looks at his figure, that Harley has no concept of common-mode signals or balanced amplification in the context of a balanced preamp.

Bob mangles power amplifiers to equally hilarious effect. We are told that an amplifier's driver also functions as a phase splitter. Now, if we were talking about quasi-complementary output stages, we would need a phase splitter, but in complementary stages one is not needed. Figure 6-4 is a complementary bipolar circuit. A common emitter stage is loaded by a resistor, and a pair of diodes in series with the transistor and resistor biases the output stage to class AB (Harley does not mention what they do at all). The output stage of Figure 6-4 is a complementary pair of bipolars. Harley tells us that "one phase [of the phase splitter] drives the PNP transistor, the other phase drives the NPN." It is clear he cannot read a schematic because his figure shows the bases of the *npn* and *pnp* shorted together by the diodes. No time-domain diagrams of the current in the output devices are shown to give the reader an understanding of class B operation.

If you want to know more, turn to 'Appendix B: Audio and Electronics Basics." To teach us amplifier circuits, he uses a bipolar device with the classic discrete biasing circuit (3 resistors and a capacitor). Below the figure that is in every electronics text, we find "Copied with permission from *Audio Technology Fundamentals* by Alan A. Cohen." It is clear that Harley is copying from this text with no knowledge of the subject. Why is he using such a complex circuit that involves stabilizing the bias current in the presence of shifts in transistor parameters? Why is he using a bipolar at all, since this introduces the complication of base current? Why can he not draw this common-as-muck circuit himself without the need to use copyrighted material?

We again see that Harley cannot read a schematic. He tells us the emitter is at ground potential, but the figure shows the emitter goes through a parallel combination of a resistor and a capacitor before it is connected to ground. The emitter is even marked 2.50 V in his figure. If the capacitor were large, the emitter would be at an ac ground, but now we are far beyond Bob Harley's knowledge and we are adding unnecessary complications for a reader whom we are trying to teach the fundamentals of amplification.

Harley then goes on to talk about bias currents (he picked a bipolar device, remember), but the figure he shows is marked *bias voltage* on the *x* axis. Cutting and pasting from other texts without any knowledge of what you are discussing causes embarrassments like this.

To explain amplification to a layman, you have to use a lot of figures to get across the concept of active amplification and the need for biasing the device. Circuit diagrams need to be as simple as possible. MOSFET or JFET devices are preferable because you do not need to deal with base current. Harley does none of this because he has not gotten to the point of understanding concepts, let alone the thorough grasp needed when trying to explain the concepts to others.

One could go on and on and on. Mistakes are repeated again and again. We are shown another complementary output stage on page 407 and told it has a phase splitter. And the big howlers keep coming. Slew rate is an input-referred specification in his eyes, and to make sure his foot is firmly inserted in his mouth he then goes on to say, "Slew rate is often referred to as an amplifier's 'speed.'" According to Harley, *npn* and *pnp* devices are called bipolar "because current can flow in two directions through the transistor." Clearly Harley has no idea about drift, diffusion, and the concept of holes. JFETs in Harley's world are quieter than bipolars and good for circuits that need lots of gain. The concepts of thermal noise, transconductance, and output conductance are clearly beyond Bob Harley. Digital circuits get half a page of discussion and no schematics. Harley says MOSFETs are sometimes used in power amplifiers. I guess he has not figured out what is used in the Pentium. He also has not figured out what a differential pair is, a PLL, or the cause of distortion. Although the names come up again and again in the text, they never get explained.

I think you get the point by now. This book lays Bob Harley's lack of training bare for all to see. Once we understand that he has no grounding in the fundamentals, we can understand how he can talk the talk of the High End so easily. It is easy to say, "Excessive feedback produces lower distortion figures, but often makes the amplifier sound worse," when you have no understanding of feedback or stability. (In the book virtually no words are spent on how feedback works, and there are no figures on the subject. It is clear he does not understand feedback.) It is easy to say a class A amplifier "generally sounds better" than class AB when your understanding of an output stage is so messed up that you see a phase splitter in every circuit. It is easy to see how someone can talk about a tubed amplifier's "spectacular soundstaging and beautiful rendering of timbre" when he hasn't the foggiest idea what the term *bipolar* refers to in a transistor.

The conclusion is simple. If you do not understand chemistry and physics, you may think you can turn lead into gold. If you do not understand electronics, you may think amps and preamps sound different and color the sound more than loudspeakers.

The Complete Guide to High-End Audio is an important document because it makes it clear that the "experts" in high-end audio are not experts at anything but B.S. and charlatanism.

* * *

Editor's Note: *David Rich puts the emphasis on Bob Harley's ignorance, rather than his True Believer's subjectivistic delusions, but rest assured, there are more of the latter in this book than in half a dozen issues of* Stereophile. *It's just that we've been over that same ground so many times.*

Tom Nousaine wants me to pass

on this suggestion to Bob and his friends: Get rid of the cheap batteries in all your remote controls. Replace them with mercury batteries. You'll be amazed how much better everything sounds!

Seriously, though, are all you peaceniks, antinegativists, and nonconfrontationists beginning to see now why "Harley bashing" and "Stereophile *bashing" aren't just a gratuitously hostile pastime? Would you have considered it possible that a 450-page book, potentially a sourcebook for future practitioners, would be lovingly prepared and published with nothing but the most egregious technical misinformation in it? Aren't you just a little bit glad that now you know what the score is? Don't you want to know about similarly flagrant instances of audio charlatanism in the future?*

Recently I gained additional insight into why Stereophile *continues to rely on such an inexcusably unqualified technical editor as Harley. I found out that a degreed E.E. with impeccable professional credentials applied to John Atkinson for a part-time job as the magazine's technical consultant and engineering "conscience." Atkinson blew him off—just wouldn't have anything to do with him. It is quite clear at this point that Atkinson doesn't want anyone on his staff who knows more than he does. Harley is a good workhorse who produces reams of technical copy; nobody seems to care that most of it is horse excrement; publisher Larry Archibald either doesn't know any better or cares only about circulation and ads; pages get filled; deadlines are met; and John remains the man who knows everything. In the immortal words of Ira Gershwin, nice work if you can get it.* •

High Definition Compact Disc Recordings
By Howard Ferstler
McFarland, 258 pages, $29.95

Reviewed by Peter Aczel

The long subtitle on this book's cover is "Sound Quality Evaluations of Over 1,400 of the Most Technically Excellent Digital Recordings." That tells you what the book is; it doesn't tell you what it is not, which may be more important to you.

It is not a music lover's guide. No music is discussed; no performance is critiqued; no pleasure other than sonic gratification is communicated. Not that I intend to hold this against Howard Ferstler. As audio writers go, he is definitely one of the good guys in the white hats—sane, accountable, basically reliable. It's just that no one, not even a freaked-out audio nut, should buy a CD strictly on the basis of sound. (Remember the Opus 3 "Test Record 1: Depth of Image"? Remember that dreadful Swedish pop song *"Tiden bara går"?* It sounded incredibly lifelike and it drove me up the wall every time I heard it as a demo. Music is not a vehicle for audio engineering. It's vice versa.)

What Ferstler does is to grade each CD as B, B+, A, or A+ in sound quality, at the conclusion of a very short audio evaluation, about 6 to 8 per page. (Obviously there are no C's and D's because those would be the ones to avoid.) In just a very few of these capsule reviews, he indicates that the CD might deserve an A++ if he were using such a rating.

The list of 1447 recordings is heavily dominated by the smaller audio-oriented labels that are generous to the audio press with review CDs (e.g., Delos, Telarc, etc.). Are these the best-sounding or just the most easily available to a reviewer? Yes and yes, at least in more instances than not.

On the whole, I have no serious disagreements with Ferstler's ratings. He knows good sound when it comes out of his speakers. The question is—can he tell good from very good and very good from extraordinary? Some of his quasi-A++ designees are the Zinman/Baltimore Berlioz *Fantastique* on Telarc, the Mehta/NY Mahler 5th on Teldec, and the Bob Mintzer "Art of the Big Band" on *dmp.* That tracks my own assessments pretty well as far as it goes (I have reviewed the latter two) but raises the question why the super recordings are all brass-heavy spectaculars? The truth is that the difference between A and A+, or A+ and A++, could be strictly a creature of Ferstler's stereo system and/or listening room. But I'm quibbling.

The book is light on voice recordings; furthermore it lists mostly single CDs, very few multidisc sets. It ignores some labels that often come up with excellent sound, such as Naxos. Its main weakness, however, is the aforesaid severance of sound from music. He doesn't even identify the conductors of the orchestras named in the reviews! A collaboration with a music critic as coauthor would have made the book considerably more valuable.

I will say this, though, for Ferstler's evaluations: he knows more about genuinely good sound on CDs than any of the music critics in the various magazines and newspapers. If all you want is a "recommended" list of recordings for sound alone, I don't know of anything better or even comparable. •

Acoustical Treatments

Echo Busters

Echo Busters, P.O. Box 247, Bethpage, NY 11714. (516) 433-6990.

Just a quick postscript. This is a company that makes acoustical treatments for home use. The prices are not at the rip-off rates some charge. The 24-inch by 52-inch panels I tested cost only $75.00 each. High-quality acoustical foam is put in a wood frame and covered with cloth. The panels are designed to be hung on the wall, or they can be freestanding with the optional stands.

This is not some tweako product that effects no measurable changes. The panels are very absorbent above 400 Hz. I deployed six of them in my room and dramatically improved the sound. They were mandatory behind the Sound Lab Quantum electrostatic speakers (not yet reviewed), and I suspect that other panel speakers would also benefit.

Placed strategically elsewhere in my overly live room, the panels enabled me to deaden it. For the price of cables that may change the sound but not improve it, you can do something that really will improve things. The only downside of the panels is that they will change the way your room looks. Here, in my bachelor pad, people say "Cool!" when they see them. Your significant other may not have the same positive reaction.

Echo Busters is run by Michael Kochman. He is very knowledgeable about room-treatment issues and is a great help in figuring where to place his products. The company also makes other room-treatment products; you may want to give them a call and discuss your needs.

—David Rich.