Issue No. 18 Spring/Summer 1992

The Audio Critic®

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The preamplifier survey, originally intended for the last issue and by now considerably expanded, is the main event. It's *the* final exam for preamps by Prof. Rich.

Your Editor reports his tests and evaluations of delta-sigma ("1-bit") CD players and D/A converters.

New staffer David Ranada begins a series of interviews with some of the deepest thinkers in audio. Part I: John Eargle, Roy Allison, Kevin Voecks, Floyd Toole.

David Ranada also shows his musicological side with a monumental *Magic Flute* review and a bit of Beethoven.

Plus the longest crank letter ever published and answered in our pages, along with sundry columns and features.

Issue No. 18 | Spring/Summer 1992

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For subscription information and rates, see inside back cover.

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Editor/Publisher' s Mutterings/Notes:

Yes, this issue is labeled Spring/Summer 1992. No, you didn't miss an issue in between; this is No. 18. Yes, it should have been the Spring 1992 issue, followed by the Summer 1992 issue. No, this won't count as a double issue against your subscription. Yes, it's almost twice as fat as earlier issues like No. 11 and No. 12, for the price of a single issue. No, that's not good business. Yes, it could have been split in two, but the first half would have appeared a little superficial, with unanswered questions. No, the Fall 1992 issue isn't scheduled to be combined with Winter 1992-93. Yes, a lot of the Fall issue is already written, so it has a good chance of being timely. No, I won't make any promises anymore. Any other questions?

* * *

This is the first issue of ***The Audio Critic*** *that isn't being sent to you by first-class mail. The latest first-class rates are absolutely unaffordable, even after the recent 9% increase in our basic domestic subscription price. No full-size magazine known to me is mailed first-class;* ***The Audio Critic*** *has been an extravagant exception. A considerable effort is being made to optimize our mailing procedures to the point where your issue will spend only a few more days in the postal pipeline than first-class material and arrive just as reliably. If there's a mailing problem in your particular case, please let us know at once. We can fix it.*

* * *

I welcome on board David Ranada, our new Contributing Editor at Large. Not many people who read audio publications are unaware of his name and previous writings. David is deeply steeped in both electronics and music, and by deeply I mean that he reads circuit schematics and orchestral scores with equal facility. He is also highly computer-literate. How many tweako journalists of the "alternative" audio press can make those claims?

* * *

Erratum: *In the "Box 978" column of Issue No. 17, I wrote in my editorial reply to one of the letters involving* Stereophile *that "I don't remember anything in their pages about the...tragic decease of the brilliant Deane Jensen..." Actually, there was a necrology (as my father would have called it) in their January 1990 issue—well over a column on page 69, by lined by Robert Harley. My apologies; as I said, I didn't remember.*

Box 978

Letters to the Editor

We get hundreds of love letters and dozens of hate letters in response to a particularly good issue, such as No. 16 and No. 17, but the ones that drive your Editor up the wall are the please-let's-have-no-more-confrontations entreaties from the knee-jerk conciliators. The basic philosophy of these kind souls is that in a shrill argument where one side screams that 2 + 2=5 and the other nastily insists that 2 + 2 = 4, the probable truth is that 2 + 2 = 4.500. That's a good working principle in family court and maybe even in politics, but not in a technological discipline. Letters printed here may or may not be excerpted at the discretion of the Editor. Ellipsis (...) indicates omission. Address all editorial correspondence to the Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951.

The Audio Critic:

Reader Barry McClune raised this point in *The Audio Critic,* Issue No. 17: If a test were published (double-blind, >95% confidence level, demonstrating some "golden-ear" type perception), what is the probability that those results would be accepted by the objectivist camp as being a valid test? Good question! Null, I expect; but first let me compliment *The Audio Critic* for printing this uncomfortable response, as the management (firmly situated in that *soi-disant* "objectivist camp") has strenuously objected to my recent research verifying a widespread "subjectivist" view. Then let me introduce myself to your audience.

For over a decade my pet topic in audio (after 78s, those Grecian urns of beautiful sound) has been Absolute Polarity. Regrettably I did not discover it myself, although by default I have become its most vociferous defender since Richard Heyser died. To many, many ears Polarity is audible even over car radios and CBs, and the necessary correction (a simple switch of wires) constitutes the first step, or at least *primo inter pares* [sic], towards real sound improvement. Hence my grand parade banner, "Better Sound for Free!" One does not often see such subversive advice, especially not from manufacturers, retailers or journalists.

By no means an equipment junkie, I advise everyone that spending *time* earns far greater rewards in audio than spending *money,* and since the gear itself has too few effective switches for Polarity, time becomes the only thing to spend to achieve this easy remedy. Why everyone hasn't already discovered for themselves this vital factor that makes reproduced music sound right astounds me. Certainly everyone who crosses my own threshold catches it! Loudspeakers are probably the principal culprits—those blatant incoherencers with drivers wired mutually out of phase to satisfy computer-designed steep-slope crossover desiderata that, ever forgetful of phase, do at least achieve flat amplitude response (on paper). Mere listeners really cannot be blamed for their oft-documented inability to discern shades of phase or even total reversal, given these conditions. Further confusion derives from the crazy variance of Polarity on discs, tapes and records—an endless admixture of in and out, once you grasp the concept.

For those unfamiliar with reality, Absolute Polarity requires the correct arrival of longitudinal transient wavefronts at the ear, replicating the waveform production of actual musical instruments, compressive mostly, as in "percussion." Electronics and loudspeakers can reverse (or mangle) that readily measurable referent to live music by creating rarefactions, thereby limiting both physical and aesthetic impact. The muffling distortion, I call it. Bass lines sound lumpy and amusical; attacks become blunted; palpability is reduced throughout the range; and speech intelligibility suffers. Over a proper minimum-phase system the effect is undeniable. Think: who could mistake a photographic negative for a positive print? And what would we call someone who does?

The auditory mechanism by which we do in fact *(pace* Helmholtz) recognize Polarity was first isolated by Charles Wood at the University of Texas in 1957, later reported in the *Journal of the Acoustical Society of America* as "the Wood effect," hence the title of my comprehensive and impolite book *The Wood Effect: Unaccounted Contributor to Error and Confusion in Acoustics and Audio* (1988). In certain locations it was very well received, notably *Audio* (heir to the Heyser legacy) and *Stereophile.* But not a word in *TAS* or *Stereo Review!* Whatever, Audio Advisor sold over 400 copies and Dr. Sao Win offered only plaudits—it must have done something right! The centerpiece was Chapter Three, in which the paper trail of Polarity was chased from its first pale glimmer at the Harvard Psychoacoustics Laboratory in 1951 up until the early 1988 deadline, with nearly one hundred printed sources, all supportive except two, cited. Peter Moncrieff: "Absolute Polarity is far more audible than most people suspect." Richard Heyser: "I propose that from this point forwards the entire audio industry take a basic step [the Polarity Convention] which is capable of improving the quality of the listening experience without adding to the cost of any product." Powerful words! Even more notable, to amateur scientists, were the double-blind tests on Polarity reported in the *Journal of the Audio Engineering Society* by Lipshitz and Vanderkooy: "...over loudspeakers, and overall, including musical excerpts, the results on the audibility of the polarity inversion of both loudspeaker channels... [represented] confidence of more than 99% in the thesis that acoustic polarity reversal is audible." *[This is correctly excerpted from the published paper, which I was fortunately able to look up; Clark Johnsen's letter arbitrarily edits these words to end up with a misquote suggesting that the authors consider absolute polarity per se to be audible.—Ed.]*

These solid results, one might think, would be difficult to dismiss; thus they serve most excellently to illustrate my argument: Here we have an audio phenomenon widely attested and proved beyond a whit of doubt statistically, yet few among us today care. Who has followed up those conclusive double-blind tests? Whither pressure from the press? Consumer demand? *Where are the objectivists now that we need them?* Ladies and gentlemen, does not their very absence here totally undermine their sanctified station? Here we recall Mr. McCIune's implication that "the objectivist camp" will dismiss any outcome contrary to the ruling paradigm, for reasons subsumed under the rubric "hidden agenda." In this case, to prevent you from free improvements!

An unsettling statement! Nor did my book, written to rectify the situation, put much of a dent in the "establishment." Professor Lipshitz himself, an erstwhile indefatigable proponent of Polarity (in *The Audio Amateur, Wireless World, Hi-Fi News & Record Review,* besides *JAES)* informed me I must conduct my own tests, more or less to make a fresh splash on the AES. That was sound advice, given after reading *The Wood Effect.* Meanwhile the book also helped me strike up an acquaintance with the venerable (and very funny) Richard A. Greiner, who also shares an enthusiasm for Polarity. He too encouraged me to "do research;" thus was born my paper for the AES 91st, "Proofs of an Absolute Polarity," in which I demonstrated *100%* confidence level in the thesis that Absolute Polarity is audible. The exact score was 22 for 22, a feat so improbable I contemplated fibbing to make it 20 for 22.

Here we interrupt the narrative briefly to describe that paper. An "Introduction" to definitions and audible aspects occupied the first two pages, "Problems Obscuring Perception" took three more, and an abridged "Survey of the Literature" was given nine. Several pages then were spent discussing blind tests in general, mostly unfavorably; this must have rankled my critics, particularly the foremost objection: "Undefined limits of resolution—no scientific evidence exists whether such procedures can actually reveal the distinctions sought. Indeed, the massive null results suggest otherwise." Think about that. "Poor selection of participants" was another, and "Side effects: Everyone knows you change when you take a test." Then came four pages devoted to "Test Design and Protocol," two of "Results and Conclusions" and three listing chronologically the fascinating printed record of Absolute Polarity. Bear these twenty-seven closely reasoned (but politically incorrect) pages in mind as you consider the knee-jerk reaction they stimulated.

I further risked an innovation dubbed Triple-Blind Testing. You demand Double-Blind? I'll see that and raise you one! Contrariwise I argued against multiple-blind testing only to earn, besides calumny in *The Audio Critic* No. 17 *[By definition, sincerity cannot be calumnious.—Ed.],* the proximate cause of this letter, a surprising commendation from Ralph Hodges in *Stereo Review,* February 1992. (On the other hand, David Moran trashed me in *Speaker Builder.* Where will it ever end?) Anyway, to resume my story, because of the announced paper yet another character entered my life, Don B. Keele, who wears among other mantles the informal title of successor to Richard Heyser at *Audio.* He wrote to request a copy of *The Wood Effect* and later regretted he hadn't noticed Polarity earlier but would mention it in future reviews. (And so he has.) Imagine then my astonishment and delight when I found myself sandwiched for Sunday morning at the AES 91st between those two fine Fellows of the AES, Greiner and Keele, each of us reporting on Polarity.

Having shared each other's thoughts and preprints, we had some good times together in the Presenters' Lounge, along with Len Feldman, Fred Davis and Corey Greenberg, but that's another story. Insider exclusive! Up on stage Don Keele exhaustively explored what he acknowledged was a previously discovered mathematical formulation (though few to this day understand), while Dick Greiner, sharing my broader brush stroke, nevertheless failed to pin down Polarity convincingly, owing only (I think) to poor choice of equipment. *[He convinced* me *alright.—Ed.]* The session (Listening Tests, Part 1) went very well. Many good questions were asked, and I was besieged afterwards for nearly an hour, although *entre nous* RAG's assured, humorous podium style made me unaccountably envious. *[Are you sure it wasn't his superior knowledge?—Ed.]* But that afternoon I sat beside him for Listening Tests, Part 2, the two of us trading observations like teenage guys at the movies, although not so loudly. *[Are you sure the buddy-buddy sentiment was mutual?—Ed.]* There, too, we shared Robert Harley's seminal (objectivists say vile) paper, "The Listeners' Manifesto," which famously split the vociferous audience. Never before (I was told) had a plenary session been called to handle overflow response. While wholly disapproving of the talk, the good Professor leaned over and vouchsafed to me, "Well, he *does seem to be* a very *intelligent* young man."

Regarding *The Audio Critic's* rather less polite dismissal of my own paper: "Pretty lightweight stuff," say they, predictably attacking the test procedure rather than caring to refute (or verify) my findings. And mistakes were made, although ye editor missed the really bad one in his drive to disallow the outcome, about which he had never himself beforehand informed his readers. *[That's #1 (explanation below).—Ed]* He would rather be cross: "...the main thrust of [Clark Johnsen's paper on absolute polarity] was how right he has always been and how wrong everybody who disagrees with him has always been." *[Again correctly excerpted here, although rewritten by Clark Johnsen in his letter.—Ed]* Peter, *give me a break!* We are *all* audio critics! Why not just have written, "YES! Here we have a novel addition to the audio armament, which regrettably I never before brought to your attention. *[That's #2.—Ed.]* While Mr. Johnsen's procedures are not unflawed, he alerts every audiophile to an elementary adjustment for better sound reproduction." But NO, nothing of the sort! What we get here, and in many publications, is denegation and ridicule *[That's #3.—Ed.]* of an actual, provable acoustic phenomenon of benefit to every listener.

For all to see, this situation demonstrates the fundamental, unspoken (until reader McClune) error of double-blind testing and why I despise same even while practicing it: Objectors in "the objectivist" camp nitpick you to death on procedural grounds whenever they wish to deny the results. Or they ignore you. Or they hurl outright insults! For example, the curious *amicus curiae* brief filed in *The Audio Critic* by one Jeff Corey, Ph.D., Professor of Clinical Psychology at C. W. Post College, Long Island University. It was a splendid tactic, introducing him shortly after the editor had lit into my "lightweight" hide. Prof. Corey possesses myriad academic qualifications, as we see, also including uncredited activity with CSICOP (Committee for the Scientific Investigation of Claims of the Paranormal), in which he takes a perhaps undue interest and where he joins hands with The Amazing Randi, prestidigitator and scientific debunker extraordinaire. Calling time out from his UFO-abductee scoping and instructional duties at LIU, this man saw fit to address the AES membership in New York City. Why? He further devoted 7½ column inches in *The Audio Critic* (out of 19) to debunking me and mine. It went like this—truth in brackets, Jeff in quotes:

"After the workshop [Cable Roast we called it, rejecting that new-agey "workshop" stuff], I was approached by a number of people who took issue with my point of view [that we were full of crap]. The first identified himself as a certified clinical psychologist [Not so; he was Dr. Michael Gindi, the psychiatrist, and we were on our way to lunch with Roger Skoff; being polite he introduced himself as a fellow psychologist, nothing clinical] and audiophile. He insisted ["invited" was the actual sense] that I must listen to his sound system [Gindi: "Have you ever heard a real high-end system?" Corey: "No."] and judge for myself. I replied that I would do so, but only in a double-blind test. [Lies! He dissimulated immediately with total lack of affect.] After we both persisted in our different views, he stalked away crying, 'And you call yourself a scientist?'" [No stalking away, no crying out loud, only a mild expression of disappointment towards one who refused to listen for himself.]

So there I was, left as the man who next "introduced himself with 'I attended Harvard and have a degree in physics. I invented the triple-blind method.'" I ask the gentle reader, do I talk that way? And do I so obviously betray my Ivy League origins? Long ago I learned that a Harvard sheepskin is no comfort to others. Nor am I myself much impressed by that woebegone, infiltrated, overrated institution. That part of my upbringing ill serves me except among the terminally credulous. Therefore I rarely trot out my degree (in Applied Physics) upon unfamiliar ground. Give me credit, please, to realize that a total stranger from LIU will not, even from his scrubland location, automatically bow and scrape, as I might be thought to wish, towards my Cantabridgian [sic] beacon. Jeff errs; I never portray myself as the Vessel of Veritas. He must have lifted it from my bio! Further indulging in (psychological?) wishful thinking, Professor Corey moots a lengthy critique of my procedures (wouldn't you know?) without evincing the slightest desire to check out my results, which now share that dubious honor with Dr. Gindi's hi-fi. Not to bore the able reader, who might well be advised to proceed to the foregone conclusion, but here follow Corey's four scholarly objections, in order, with rebuttals, to my "hopelessly flawed" study.

"1. No informed consent was obtained." What federal, state or local statute requires *that!* For *listening tests?* God, let's hope not! Besides, that would invalidate the whole triple-blind protocol. *This* is the *numero uno* flaw?

"2. Reverse polarity was always presented first." True, and the next run shall switch the order, and the third offer only invariant couplings. As with color separation prints, the plan adds up to more than the apparent sum of black-and-white parts. Pity mention of this was omitted in the paper that became prolegomenon to a work in progress; at least my head was in the right place, if not my pen! HOWEVER, all may be forgiven because of an exceptional circumstance: One of the choices was correct *a priori.* I'll explain. Blind tests in audio are usually employed to make judgement calls or establish rankings on equipment; but here, uniquely, the procedure was performed *on* the subjects rather than *by* them. Like multiple-choice exams, the right answer remained just that regardless of its position on the list, obviating the need for rigorous presentation modality *à le Nousaine* [sic]. (People notoriously prefer to say they heard a difference.) Subjects here not only voted for an audible differentiation between one sample and the next, but unanimously made the correct call for right Polarity *and* freely commented on how much better it sounded! No other test in audio, blind or not, to my knowledge, has legitimately bypassed the obligatory regimen—which shall nevertheless be fulfilled. Apart from that, the protocol was far more skillfully designed than *The Audio Critic* acknowledged. To quote a desideratum from Floyd E. Toole's *Subjective Evaluation,* "the [dire] effects of group voting [were] eliminated by using single listeners," a tiresome procedure indeed. In fact nearly every criterion in his cited essay on audio testing was faithfully observed. Moreover, several subtler aspects were attempted beyond Toole's declared scope, All these were disregarded by Jeff Corey in his rush to judge me.

"3. The operator always knows the condition being presented." And how could he not, unless deaf, since Polarity (as proved) is perfectly audible? To Prof. Corey the operator's innate ability becomes yet another procedural objection. Even "being away during the second presentation does not count," he avers, referring to my honest attempt to keep the subjects blind as possible. Just what does he meat by that? Was knowledge somehow communicated through the wall by, what, psychic vibrations? Call the CSI-COPs!

"4. There is no guarantee that the procedure of reversing polarity *[Clark Johnsen inaccurately quotes "that reversal of Polarity"—Ed.]* did not affect other salient features of the presentation *[Clark Johnsen omits "of the presentation"—Ed],* such as the exact decibel level." Jeff skates on *mighty thin ice* here. Fact is, switching cartridge or loudspeaker leads (or equivalent operation in the digital domain) to make the Polarity change has *zero result on volume.* This objection goes spectacularly awry, proving his awesome unawareness of Polarity and more.

Dr. Corey concludes, "While polarity may have a real effect," (thank you, thank you) "this study did not demonstrate anything warranting the author's conclusions;" which is strange to hear, as the principal conclusion was that Polarity has a real effect, one very palpable to every testee, all of whom testified to same in their written commentary. No matter. Surpassing himself, Jeff derides my effort as "a hoax" and likens me to "a blind wombat." *[He called your triple-blind design, not your total effort, a hoax, and you obviously "didn't get" the quoted Monty Python joke about a blind wombat.—Ed.]* Such colorful language! But what place has name-calling in a journal of rational discourse? Especially since the recipient of those epithets had only attempted to reconfirm, for younger ears, the express views of Hansen, Madsen, Lipshitz, Heyser, Peters, Butt, and Meyer in *JAES* over a decade ago. And many others in other locations, most decisively David Stodolsky in the September 1970 *IEEE Transactions on Audio.* Jeff may not have known the long history of this idea, but does that excuse his conduct?

If any fraud has been committed, it is on innocent listeners who have been, shall we say, sheltered from the reality of Absolute Polarity. Which means, to repeat: Initial transient wavefronts must be reproduced as such and not be made into rarefactions, for the feel of real musical instruments to be recreated. Ladies and gentlemen, think: If I am correct, you have been systematically misdirected from Better Sound for Free by "experts" and reviewers left and right, over- and underground. But if I be wrong, then Harry Pearson, Peter Aczel, J. Gordon Holt, and Julian Hirsch may raise a hearty toast against yours truly, since none of them has ever uttered a nice word for Polarity. *[That's #4.—Ed]* Picture it! That odd quadrumvirate locked in mirthless embrace.

Anyone want some more proof? Let me quote a good one straight from *The Wood Effect:* "Finally, reassuringly, the list named Baker's Dozen Best-Sounding Records from *The Absolute Sound* circa 1985: [Here follows the thirteen by name, all save one shown to have the exact same Polarity.] The splendid consistency of that list, where no one contrived, wished or anticipated it, sublimely and even *double-blindedly* illustrates the key role Absolute Polarity plays in making sound decisions... Almost without exception, everyone's favorite records match the fixed polarity of their systems."

My best advice: Listen for yourself, never mind the media. And do this test: Choose a cleanly recorded 30-second segment of music and audition it two or three times, then switch all wires on both speakers and listen once or twice again—for variant impact and musical intelligibility, especially in the bass. Then switch back and repeat. Warning: Simple 6 dB/octave crossovers reveal the effect best and LPs outdo CDs. If you still have trouble, send $4 for AES Preprint 3169 (K-3) or purchase *The Wood Effect.* Should this letter encourage anyone to experiment and play, the ribbon and ink shall not have been wasted.

Clark Johnsen
The Listening Studio
Boston, MA

I'm taking my cue from your cover letter to the above, in which you wrote:

"Dear Peter—This letter will be a toughie for you to print. In fact, maybe you should rip it up right now! I can't think why I spent time on it, but it's yours now. Enjoy!"

Well, I have no problem at all printing it (except for its inordinate length, but that comes with the territory) and I'm certainly enjoying it because one of my favorite cruel pastimes is balloon puncturing especially when the balloon is filled with self-congratulatory gas.

Might as well begin with the biggie toughie—for you, that is. In four different places above, which I've numbered in brackets—once is not enough to drive home your strongest point, right?—you put down this Editor and his publication for having ignored the subject of polarity. (Unlike you, I refuse to capitalize the word lest it should be taken to mean "the quality or condition of being Polish.") Well, take a good look at the sidebar on this page. That was written more than 13 years ago, long before **you** ***mounted your "polaritarian" soapbox, amigo. And it's considerably more succinct, lucid and impartial than any of your windy variations on the same theme (at least in my modest opinion). How is it possible that you just*** **knew** ***that I had never alerted my readers to these issues when at the same time you list in the appendix to your paper every little tweako hiccup on***

"A Brief Note on Absolute Phase"
(Reprinted from the Winter/Spring 1979 Issue of *The Audio Critic)*

When a trumpeter at a recording session blows into his mouthpiece, the first transient wave front emerging from the bell of his instrument, the initial attack, pushes the microphone diaphragm in. It's a positive-going signal and should be reproduced by a loudspeaker diaphragm moving toward the listener—a push. Similarly, a singer taking a sharp breath initially sucks the microphone diaphragm out and creates a negative-going transient signal that should be reproduced by a pull of the speaker diaphragm. If these signals are reversed in polarity, making the speaker push when it should pull and vice versa, the perceived sound won't be exactly the same. There will be a subtle loss of realism.

The audibility of "absolute phase" in music (not to be confused with stereo channel phasing!) has been known for a long time to sophisticated audio practitioners; in fact in the early vacuum-tube days it was an ironclad rule in the recording studio that there must be no phase-inverting stages anywhere in the recording and playback chain. This traditional piece of studio wisdom is now being rediscovered with wide-eyed wonder by assorted new audio gurus and cultists, who hail it as the invention of the wheel.

With the widespread use of multimike, multichannel, op-amp-console, mixed-down recording, the absolute-phase criterion has become meaningless. Not even the cleverest recording engineer knows what happens to a positive-going pulse through that maze of signal paths; even if he did, he might end up mixing his signal with inverted versions of itself on the same track.

With exceedingly simple recording techniques, however, such as are used by Mark Levinson, Proprius, the "new" Max Wilcox and a few others, there remains the possibility that the positive or negative-going character of a signal will be preserved intact. In that case an extra touch of realism can be added to the reproduction by experimenting with the plus-minus polarity of each channel, either by quickly reversing the speaker leads on each side by hand or having some kind of two-position switch in each channel. (Needless to say, it won't work with speaker systems that have the woofer pulling when the tweeter is pushing—or have any other driver out of phase.)

Try it. You'll hear it. The better-sounding of the two possible connections will be the one with absolute phase.

* * *

If I ***were writing the above today, I would qualify and circumscribe the audibility of absolute phase even more carefully because when I press the Invert button on my current reference preamp I usually hear no difference whatsoever. I have been able to hear*** **something** ***on rare occasions, however, especially through coherent electrostatic loudspeakers (such as Quad ESL-63's). That something is never really dramatic or thrilling; "Better Sound for Free! " promises far too much. —Ed.***

the subject by certified ignoramuses like Neil Levenson and Enid Lumley? I think "lightweight" is an extremely mild label for such a mentality.

Unless you're totally shameless, you should now be looking for a suitable hole to crawl into and disappear, but I can't let you go just yet. You'll have to sit still while I point out some of the other "hopelessly flawed" aspects of your massive exercise in self-discrediting (although I must leave a few for Professor Corey).

First of all, why do you think that anyone who rejects your methods and procedures also rejects the idea that polarity may be audible? Are you the icon, the totem, of polarity? "He that is not with me is against polarity." Such conceit! Believe me, it's possible to be of the opinion that polarity inversion is audible under certain circumstances and that Clark Johnsen nevertheless doesn 't know which end is up. You certainly don't seem to have the slightest inkling of that elementary concept of logic called petitio principii, *dealing with the fallacy of taking for granted up front the truth of something that is to be proved. Your "rebuttal" of Jeff Corey's point #2 (a priori, a schmiori) completely violates this ground rule of clear thinking. If you can't see that, then you have no business writing AES papers and making demands on the valuable time of professionals. As for Dr. Corey's point #4, "quickly switching wires "—your own words in your paper—can indeed cause level differences of 0.2 dB or more, unless the switching system makes perfect contact every time. Switching by hand could be iffy.*

A couple more condiments for the humble pie you'11 have to eat:

Primus inter pares *is the correct phrase. The adverbial* primo *doesn't fit into your sentence structure. The Medieval Latin name of Cambridge was* Cantabrigia, *and the correct English proper noun or adjective is* Cantabrigian, *without a* d. *Any native or resident of Cambridge, Massachusetts, can also call himself/herself Cantabrigian, but a Latin-spouler from the Harvard campus is expected to know how to spell it. And then,* à le *doesn't exist in French. It's either* à la *(feminine) or* au *(masculine) or* à 1' *(if the noun begins with a vowel) or* aux *(plural).* Entre nous, *Clark old buddy, your* soi-disant *French sucks. It would be better if you concentrated on making scientific sense at all times instead of ineptly cultivating literary/cosmopolitan mannerisms.*

I'll leave the last word to Jeff Corey, but let me warn you before I sign off that I don't intend to publish another long-winded, self-indulgent letter from you. You've had your open-ended soapbox opportunity. If you still have something to say at this point, keep it short and relevant.

—Ed.

Prof. Corey replies to Clark Johnsen:

Apparently Mr. Johnsen has as much difficulty researching facts and recalling conversations as he has in understanding how to design listening tests. I am an experimental, not a clinical, psychologist. I am not a member of CSICOP and have never "joined hands" with James Randi or indulged in "UFO-abductee scoping." (Sound kinky, though. Do I use the proctoscope that I used to read Johnsen's screed? *Merde alors!)*

The conversations with Johnsen and others were reported accurately, edited to eliminate long-windedness only. However, the real problem remains that Johnsen and others of his ilk seek to scam us with hoaxes like the "triple-blind" and "subjective" listening tests. They would like us to believe that we must pay attention to the results of their badly designed tests. That is like asking us to consider seriously the results of a drug study with no placebo control or protection from experimenter bias. It's a good thing that the "subjectivists" and their toadies do not run the ethical drug companies.

Jeff Corey, Ph.D.
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C. W. Post College
Long Island University, NY

The Audio Critic:

Readers of *The Audio Critic* may have been misled by the "There You Go Again!" postscript on p. 46 of Issue No.17, in which Mr. Aczel stated that he used a "commercially available list" for the promotional mailing made on behalf of *The Audio Critic* in the Fall of 1991, a list "that had been routinely rented in the open market." Nevertheless, this mailing did appear to have been specifically sent to *Stereophile's* subscription list; I reiterate that if this was the case, it was done without *Stereophile's* permission or knowledge. Mr. Aczel has assured *Stereophile* that he acted in good faith and that he will not use this list again.

Readers may also have inadvertently been given a wrong impression on p. 45 of the same issue of *The Audio Critic.* Mr. Aczel implied that he couldn't understand why *Stereophile* had wanted Contributing Editor Corey Greenberg, rather than Technical Editor Tom Norton or myself, to take part in Dan Dugan's cable workshop seminar at the 91st AES Convention. For the record, while I understand that Mr. Dugan had at one time thought about asking me to take part, he never actually did so. Instead, at what I believe to be the suggestion of their mutual friend, Ken Kantor, he asked Mr. Greenberg.

Thank you for allowing me this opportunity to put the record straight.

John Atkinson
Editor, *Stereophile*
Santa Fe, NM

There you go again—and again! You are not putting the record straight. You are just muddying the waters. A mailing list is either available for rent in the open market or it isn't. If it is, then no "permission" is required. (Anyone with a credit card can rent a Hertz car, whether or not he is driving—let us say—to a meeting to organize a hostile takeover of Hertz.) If you want to screen the renters, that's your lookout, not theirs. Your "nevertheless" is just double-talk

What happened was that I received a threatening letter from Stereophile's *lawyer, who didn't have a leg to stand on. I pointed out the facts of the case in my reply and I haven't heard from him since. In that letter I did say that the list will not be used again—my promotional plans don't call for it—but I "promised" nothing. I don't report to* Stereophile *for business decisions.*

As for the AES convention, Bob Harley actually had to take a lot more heat than Corey Greenberg, and what I was really "implying" was that you, primarily, and Tom Norton, secondarily, should have placed yourselves in the hot seat to defend Stereophile's *views, regardless of which session or workshop was involved. That's what I would have done in a similar situation, but then my views are somewhat more defensible. Dan Dugan tells me that he made repeated attempts to invite you but you never returned his messages.*

Now the record is straight.

-Ed.

The Audio Critic:

Thank you very much for the splendid and fair review of our Q/4 loudspeaker, your Issue No. 17, p. 11.

I draw your attention to the second paragraph of that review and the statement, "which has no provisions to minimize diffraction."

In point of fact, the Q/4 comes with our "antidiffraction grille": the grille frame has a bevel from front panel outward, the bevel serving as a conductor of the energy flowing along the front panel outward, rather than against a square corner.

Proof that this works is the fact that the loudspeaker sounds smoother and larger with the grilles in place, contrary to the usual moderately priced loudspeaker, which sounds better with the grilles removed.

You rightly point to the "remarkable" bass of the Q/4. We have measured response down only a few dB at 35 Hz in most rooms, and you have identified the reasons—the "line tunnel" loading produces highly damped bass with approximately 12 dB/octave rolloff anechoically, and thus permitting "room gain" to flatten the response well below the anechoic cutoff of the system.

Again, thank you for the review; I hope the clarification above will help your readers and the general cause of truth in the marketplace of high fidelity.

Sincerely,
Irving M. Fried
Fried Products Company
Conshohocken, PA

I didn't notice your marginal little grille tweak and I no longer have your speakers to check it out, but I have no reason to contradict you.

Room gain is another matter; it's rather unpredictable and in a large room like mine (almost 4000 cubic feet) of very small consequence. The only fair and practical way to compare the bass response of speaker A with that of speaker B is anechoically (i.e., by measuring the extreme nearfield response), although some very credible authorities like Roy Allison might disagree with that statement.

I say, give me the best possible bass response anechoically, and I'll worry about the room later. That philosophy has never let me down.

—Ed

The Audio Critic:

.. .[In] Issue No. 16 of *The Audio Critic,* I was enjoying your no-nonsense approach to the evaluation of audio equipment until I reached the MSB Technology Corporation advertisement. After reading on page 57 about your irritation at having allowed Music Interface Technologies to place one of their snake-oil speaker-cable ads in your publication, I was disappointed to find magnetic and mechanical snake-oil being promoted on page 34.

A single plate of *any* material does *not* provide magnetic shielding to adjacent equipment. The Isolation Plate may be a magnetic barrier, but anyone claiming that stereo equipment will perform better when placed on top of this plate as a result of the plate's magnetic shielding is either ignorant or dishonest. Even if the plate were formed into a complete enclosure, what puts out a magnetic field strong enough to couple into the circuitry within audio equipment? Nothing normally found in the vicinity of stereo equipment.

I'm not going to bother arguing about the vibration-damping claim they also make. It may make a difference for your turntable if you live next to railroad tracks, but who else benefits? For a CD player, is this claim any different from those of the CD rings and clamps? As for vibration damping being able to improve the performance of an amplifier, preamp or any other nonmechanical component, just give me a ticket on the first train back out of the Twilight Zone.

Please review the ads in your publication more carefully in the future.

Best regards,
Graham Ross
San Mateo, CA

Everything you say is 100% true to the best of my understanding. **Mea culpa.** ***The only excuse I can offer is that the ad was a last-minute insertion, just before press time. It's a weak excuse, but Mark S. Brasfield is known to me as a highly qualified and knowledgeable technologist, and thus my tweako warning light didn't come on. What can I say? At least the ad wasn't repeated in Issue No. 17. The life of a no-nonsense editor—thank you!—is not an easy one.***

-Ed.

The Audio Critic:

I felt compelled to write again, to set the record straight on several aspects of my letter and your response [Issue No. 17, pp. 3-4]. Like many degreed engineers, I do not have the literary prowess of someone like yourself. That fact, plus the subtleness with which I attempted to chide you, resulted in your missing my point. As you stated, there was a point where I ran out of references. There were reasons why the first six paragraphs were supported by the appropriate technical references and why the seventh paragaph seemed to come out of nowhere. That paragraph was meant as a chide *[sic]* of your text in a previous article, before you had started the migration from, as you put it, "tweako subjectivism to the land of science." I probably should have referenced that seventh paragaph as follows: [8] Peter Aczel, "Speaker Wires and Audio Cables: Separating the Sense from the Nonsense," *The Audio Critic* 2.2 (1979): 23-26. Excerpts from that text follow: "The realistic criteria. ...A good dielectric, such as Teflon, is also an important requirement in a quality audio cable; dielectric materials chosen with cheap and easy fabrication in mind often exhibit capacitance changes with varying signal frequency and voltage, which may in extreme cases be the cause of spurious modulations of the signal.... The most farfetched idea about speaker wire performance comes to us from France. It calls attention to the possibility that the distance between the plus and minus leads will be minutely varied by the magnetic field between the two wires as well as by the acoustical energy in the listening room [microphonics]. This would cause a fluctuation of the energy storage in the speaker leads and thereby modulate the audio signal. Wild, isn't it—but not completely without plausibility, especially at current levels of several amperes, which are quite common in loud playback through large amplifiers. According to this theory, very rigid speaker cable with solid (unstranded) wires will minimize the effect. We have absolutely no opinion on the subject but are willing to concede that this kind of undesirable modulation might be marginally *audible* [the emphasis is mine] under worst-case conditions (such as the third round of Pernod without water)." *[Mr. Mohler conveniently leaves out that last qualification in parentheses.—Ed.]* "But 'dielectric issues' [materials]? Vibration? Microphonics?" Well, Mr. Editor, it is you who used to make the journey into Enid Lumley's nonexclusive cloud-cuckoo-land. By the way, one should be careful not to insult electrical engineers with three college degees when one is in fact criticizing one's own writing, albeit poorly presented on my part. The reasons for my chide *[sic],* which I kept to myself in my last letter, are as follows:

1. When one grows substantially, technically, as you have, one should not be so quick to publicly cite the shortcomings of other audio periodical writers who may still be growing. Disparagement of others does not make interesting reading or increase either your technical or literary position in the industry. I was trying to subtly point this out; my apologies—I should have been clearer but I suspected that if I were, the original letter would not be published, as I suspect this follow-up letter won't be. The weight of the BS prevalent in some magazines will be their own downfall without

anyone else citing examples.

2. Exaggerated comments like your "you could use a wire coat hanger with the contact points scraped" do not lend strength to your arguments and raise serious credibility issues. The loop area of two wire coat hangers and the higher R and L contradict some part of each of our points. Oversimplification of technical issues and condemnation of all expensive and/or complex cables is only valid if one is trying to save money rather than spend a few dollars to ensure that cables are not a problem. Exaggerations and incomplete analysis of the subject are why I wrote the first letter—and no, I don't feel that your article covers the "measured performance" issues and the "valid criteria." You focused almost solely on amplitude response variations. For example, the issue of IIM distortion is not discussed in either your or Dr. Greiner's article, and yet you say in your reply you totally agree with it. Response variations are trivial to minimize, whereas some of the other technical issues in my first letter are not. While I do not agree with their theories on interconnect or speaker cable, some manufacturers who spend a great deal of money on equipment and R & D have been blasted in your periodical. Again, I would not be too quick to criticize; at least some cable companies are making measurements!

3. As for your small chide *[sic]* of my "very slight commercial interest" in audio cable, I made a few hundred dollars in over two years of consulting for several leading recording companies (Dorian for example) and audiophiles with top-notch systems. A few hundred dollars profit over two years' time, after many hours of research and thousands of dollars of test equipment, doesn't give me too big an axe to grind in the area of biases.

So, I score my first letter 75% solid science, 15% legitimate technical concerns (not covered in your or Dr. Greiner's articles), and 10% ineffective chide *[sic]*.

Since I have chosen to write again, I would like to make several additional points. *[Some of the latter were judged to be of limited editorial interest and are omitted here to save space.—Ed]* ...You make no mention of the issues associated with shields possessing apertures in them, compared with solid or rigid shields or the materials used for those shields [1], [2]. Also, while I agree with your three points of good interconnect cable design, item three is not as you put it in your subhead a "ridiculously simple subject." Shields that preserve the signal integrity of, say, a well designed CD player that is yielding a -98 dB signal-to-noise + distortion (full scale) performance is extremely difficult. Note that the majority of commercially available cables will not perform this well even under relatively low-noise field conditions. Line-level interconnects that will perform better than the -98 dB level for electric and magnetic fields are neither easy nor inexpensive to design, even if one adheres to standards like pro audio uses: differentially balanced cables, inputs, and outputs, which are only now becoming popular in consumer audio [3]. An analog cable for microphones that can perform at a 20-bit level with respect to noise, in order to preserve the fidelity of a 20-bit digital recording prior to the A/D stage, requires rigid quadrax or rigid balanced cables utilizing mu-metal/copper tube shields. This is a requirement of one of my recording-studio customers. One can argue that, since the dynamic range of a listener's room is much less than this, performance at this level is unnecessary except as an exercise in engineering; however, I prefer to let my customers decide what they want to worry about. Another subject almost never discussed is the issue of how different components interact with respect to stability, grounding, etc., when connected via a common interconnect cable. If you believe this is a ridiculously simple subject, or inaudible, I would encourage more testing/reading/thought on your part [4].

Sincerely yours,
David S. Mohler
Westminster, CO

References

[1] H. W. Ott, *Noise Reduction Techniques in Electronic Systems,* John Wiley & Sons, 1976 (copyright AT&T Bell Labs), 50-51.

[2] Ibid., 137-70.

[3] Ibid., 40-46.

[4] Idid., 54-90.

Like your original letter, this one fires a lot of shots in a lot of directions, so to keep things straight I'll have to respond to each point in the order in which they appear in the letter rather than in the order of their importance.

• *You're far too modest about your "literary prowess." I love your delightfully archaic use of "chide" as a noun! (No, it's not at all ungrammatical, just not of this century.)*

• / *don't for a moment believe that you were being ironic when you proceeded to list the tweako cable criteria. Anyone with that kind of irony in mind, no matter how lacking in literary prowess, would have alluded just a little more pointedly and specifically to that almost 13-year-old half-forgotten piece of writing. Instead, you went into specific details of your own agenda—Kapton, polypropylene, cable jackets, etc.—which were definitely not part of the old article. Subtle chide, forsooth! You made some poorly considered statements that you were unwilling to defend when challenged and this is how you 're trying to wriggle out of them.*

• *I admit, of course, that the old article wasn't totally free of tweako influences, although it had been conceived and written as an antidote to cable cultism and was fiercely denounced by the tweako community. My present perspective and methods had not yet fully crystallized at the time, and I still held some wide-eyed audiophile beliefs. You'll have to admit, even so, that the tweako stuff in the article was heavily qualified ("in extreme cases, " the Pernod bit, etc.) and that some of the funky tube preamps of the day, with their very high output impedances, may even have constituted an extreme case now and then. None of that is an excuse, however; I was a borderline tweak then and I am not one now; I was wrong then and now I know better. The truth is more important to me than saving face.*

• *If you felt insulted I'm truly sorry; that was definitely not my intention. It's quite clear that we're allies, not adversaries, in most audio controversies, and we shouldn't allow our slight differences to get out of hand.*

• *Unlike you and assorted other non-confrontationists, I firmly believe that those who publicly/aggressively/arrogantly proclaim that 2 + 2 = 5 should be quickly and severely humiliated. No, you're wrong; their downfall can take forever if they aren't exposed and ridiculed—I've said this so many times—and meanwhile they pick up followers. I don't care if they're still growing; let them grow behind closed doors at their own expense, not the audiophile community's. That goes for me, too, in my 1977-1980 phase; maybe somebody with superior knowledge should have kicked my butt if indeed I had overstepped the line of sound science.*

• *As you can see, informed rational, and interesting criticism, no matter how damaging, doesn't end up in the Editor's wastebasket at The Audio Critic.*

• *Come on, the wire coat hanger was hyperbole, and you know it.*

• *Yes, of course, I bow to the Otala, Cordell, Cherry, etc., references in your* *first letter, but come on, the cable issue had little or nothing to do with the identification and analysis of IIM. Furthermore, I never claimed that my layman-oriented commentary on wires and cables constituted a complete treatise on the subject; on the contrary, I carefully pointed out the limited nature of my contribution. Don't expect me to be the AES and IEEE rolled into one.*

• ***This is important enough to be printed in boldface and is also a response to a number of letters not published here for lack of space (from Charlie Hand of San Jose, CA; Floyd R. Martin of Santa Ana, CA; possibly others not on my desk as I write this): Any effect whatsoever that a length of cable may have on the signal it is transmitting is measurable in a test—*** **some** ***kind of test—using sine waves as the test signal. I'll go to the barricades with that statement; there are no exceptions. People who claim to hear the exceptions are not conducting the listening tests properly; it always turns out that they have failed to eliminate observer bias and the placebo effect, or that there are measurable effects of some sort after all.***

• *You're fully entitled to your "very slight commercial interest" and I'm sorry that you felt I was giving you a hard time about it.*

• *How you "score" your first letter is entirely your business, but you're not the one to whom it was addressed for an editorial reaction.*

• *This next item was something that activated my tweako warning light—that business about the great difficulty of -98 dB shielding at line level. Huh? I haven't checked your H W. Ott reference, so I don't whether he supports you on this or not, but I can report a quick little experiment that was my response to the warning light. The cables I use with the Audio Precision measurement system are garden-variety Canare L-4E6S microphone cables. They have high-density braided shields and are very reasonably priced. I took a 6-foot length—female XLR, shield floating, to male XLR, shield grounded—and connected the generator output directly to the analyzer input, allowing the cable to dangle near the computer's color monitor. Using 2 V rms (the standard CD full-scale output level) as my reference, I obtained THD-plus-noise readings of -112 dB from 20 Hz to 2 kHz and -107 dB from 2 kHz to 20 kHz. That was with a 22 Hz to 22 kHz bandwidth limiting filter on the analyzer; expanding that to 80 kHz made the measurements worse by 2 to 5 dB, depending on frequency. Absolute worst case: -102 dB at 20 kHz, with all harmonics up to 80 kHz included I rest my case.*

• *Now, at signal levels straight out of the microphone, with 20-bit digital recording—hey, there you have a shielding problem, no question about it. We're talking about -122 dB and a very fragile signal. My article, however, was about line-level interconnects for home audio, not about solving every remaining problem in professional audio engineering.*

• *You're quite obviously one of the good guys in the white hats, audiowise, so let's part as friends. As a missionary once said to another missionary of a slightly different denomination, "After all, we both serve the same Master—you in your way and I in His."*

—Ed.

The Audio Critic:

At an early age we learned that the performance of any chain could not be expected to exceed that of the weakest link. It is gratifying to note that recognition of the weakest link in the audio chain is coming to the fore and eventually might be allotted as much time and space as the great cable controversy. I speak, of course, of a truly crucial interconnect, that which connects the ears to the speaker system—the listening room.

Speakers have been developed which exploit room reflections, with designations like "Stereo Everywhere" and "Concert Hall Sound." Also, with electronic synthesis one can miraculously turn the room into a cathedral or even the Super Dome if desired. Despite all this, there remains a hard core, dedicated to recreating an accurate sonic copy of some source—it is called "High Fidelity."

I suspect many incurables have stretched their pocketbooks to acquire an impressive array of high-quality components, then without any qualms proceeded to dump the output into an acoustically unknown quagmire. Having done it and seen it done for years, it seemed like par for the course. In 1986 I came of age sonically, called a halt to such nonsense, and set about cleaning up my own particular quagmire.

A system not requiring attachment of any materials to the walls was devised as outlined in referenced *Speaker Builder* articles. This can be expanded to near-anechoic conditions in almost any room. In my own case, I experienced for the first time a feeling of a direct pipeline to the source—an often longed-for "listen through" sensation. With it came the realization that transient behavior was the greatest benefactor in a controlled room.

My room is specifically intended for audio hobbying and aesthetically speaking could be in trouble for those who like to look at their music as well as listen to it. A recent *(Stereophile,* October 1991) article seems long on rhetoric and decor but short on acoustic control. What is needed is a *reliable* look at commercial products.

Who better than *The Audio Critic* with their already specific look at the bass end (Issue No. 13)?

Donald F. Scott
Houston, TX

References [all authored by D. F. Scott]

"Repressed Reflections," SB Mailbox, *Speaker Builder* 10.3 (May 1989): 61-62.

"The Big Box," SB Mailbox, *Speaker Builder* 10.6 (November 1989): 70-71.

"Speaker-to-Ear Interface," *Speaker Builder* 12.5 (October 1991): 44-45.

Room treatment is a big, important subject and definitely deserving of an in-depth discussion in The Audio Critic. Rest assured there will be one in the not too distant future. (Rest further assured I don't rank "the great cable controversy" any higher in importance—sonic importance, that is—than you do.)

Not everyone, however, shares your enthusiasm about a highly nonreflective listening environment. There are pluses and minuses; every situation is pretty much a law unto itself. If everybody did it your way—a very good way, in my opinion—not everybody would be as happy with the results as you are. Some people prefer quite a bit of echo and slightly smeary transients, but then some people think the food tastes wonderful at McDonald's and Burger King.

—Ed

The Audio Critic:

With regard to blind testing, I've always found it amusing that the "golden-eared" subjectivists can't hear so well without using their eyes! These are the same people that regard the ear as the only valid instrument with which to evaluate audio components. I would have thought that hearing sensitivity *improved* in the absence of other sensory inputs. A blind audiophile would be an interesting case study, don't you think?

Mark L. Swierczek
Great Mills, MD

Touche"! (Let Ray Charles identify the right one, baby, 13 out of 16 tries!)

—Ed.

Reasonably Priced Preamplifiers for the Reasonable Audiophile

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

How good does a preamplifier have to be and how much does it have to cost? According to this survey of current models, even the most fastidious audiophile can find happiness for under $800.

"Oh, you're testing cheap preamps," said a high-end dealer when I told him which preamps I was testing for this article. These units are nearly all in the $600 to $800 range. That is not "cheap" to me and, as we shall see, there is in fact little reason to spend any more than that (except perhaps under certain unusual circumstances).

This survey was inspired by my experience with my Tandberg 3002A preamp, which I purchased a number of years ago. I had then wanted to replace my Audire Legato preamp (reviewed by the Editor in Volume 2, Number 2, i.e., Issue No. 8) with something more high-tech. I chose the Tandberg after carefully researching the circuit topologies of the preamps of the period and comparing their performance specifications. To my surprise, after only three years all the controls of the Tandberg had become noisy and the function selector became intermittent. When I discussed this problem with a repair technician, he was not surprised. "Tandberg always uses cheap parts" was his reply. Despite my care examining other aspects of preamp design, I had not even considered parts quality when I chose the Tandberg. In this survey I will not make that mistake again. I will, of course, also examine circuit topologies and test measurements, but the emphasis will on "build." You will see that price and quality do not always correlate. My inexpensive Audire is still working just fine, fifteen years after I bought it.

I tried to assemble as comprehensive a set of preamps as possible. Only Adcom and Counterpoint refused to participate. Through a friend, we borrowed a brand-new Adcom GFP-565 from a dealer's stock for this survey. Counterpoint originally promised to send the Solid 8 preamp but only on the condition that it would not be reviewed with any other preamp in the same article. The fact that they are not confident enough in the product to have it compared directly with others should speak for itself.

One final note before we begin. I did not look at any tube preamps for the same reason that *Car and Driver* is not reporting on buggy whips. *[The prof said it, not I.—Ed]*

Listening Tests

Each of the preamps' line stages had ruler-flat frequency response as well as vanishingly low noise and distortion. That being the case, you would expect that the line stages would sound the same under double-blind conditions, and that is just what happened. But the important point is not how they sounded on Day One but how they will sound on Day 1000 or Day 10,000. (Yes, audiophiles—*music lovers* often keep equipment over 20 years.) Over the long haul the quality of the switches and potentiometers will determine the sound quality of a preamp; poor-quality ones will give themselves away by noise and distortion. I am sure I could pick out my not so old and not very heavily used Tandberg on an ABX test 16 out of 16 times. As we shall see below, it is not likely that any sonic degradation will occur in a 20-year old Bryston.

Phono preamp stages are more difficult to evaluate. Our measurements again indicated that no differences should be heard except for frequency-response errors. But you cannot separate frequency-response errors from other effects in a listening test of phono preamp stages. Another problem is that it is impossible to use basic double-blind comparison techniques to evaluate the phono stages because you can't parallel a phono cartridge across two preamplifier inputs and still expect the cartridge to operate correctly. Thus, for an ABX listening test, both the inputs and outputs of the phono stage must be switched. Switching low-level phono signals without distortion and telltale switching noises can be very difficult. In addition, the phono stage will experience a large transient signal during the switching process, and significant settling time is required for the DC bias signals in a phono preamp to return to normal levels. This time can be especially long for preamps using DC servo circuits with very long time constants. The last problem makes rapid switching between DUTs [devices under test] impossible.

The significant deficiencies of the obsolete vinyl medium made level matching between preamps when using a turntable and a test disc difficult. Surface noise, speed variations, and amplitude variation due to the test record's being nonplanar resulted in a significant variation (0.5 dB) in the level of the test tone. This made level matching to 0.1 dB impossible. Level matching could be accomplished by injecting a test tone directly into the phono stage, but level mismatches due to cartridge/preamp interactions would not be accounted for if this method were used.

Even if these problems could be overcome, the original problem remains: phono stages can have frequency response deviations large enough to make the stages distinguishable. Thus a positive ABX test result would probably be explained by the presence of these frequency response deviations.

I attempted nonblind listening tests of each phono stage with levels matched as closely as possible. My record collection contains a number of excellent-sounding discs which are no longer in print and have not been issued on CD. (No, I am not going to tell what discs I used because that would only result in increasing these records' prices in the Ars Antiqua used-record list. All the records I used have significant musical value, and I much prefer that any available copies be purchased by true music lovers and not some overzealous audiophile.) I noted very small sonic differences in my nonblind tests. These differences were most likely due to RIAA equalization errors, cartridge/preamp impedance interactions and/or differences in preamp noise levels. A very small chance exists that the differences are the result of dynamic stability problems (these will be discussed below). The differences could also have been the result of residual level differences or just a product of my imagination. In any case these differences were so small that I developed no distinct preference for any of the phono stages. Any of these preamps has electrical performance which far exceeds the limitations of the vinyl medium—such as poor signal-to-noise ratios (surface noise), very high impulse noise (clicks and pops), large frequency-response errors from the cartridge and, worst of all, inner-grove distortion (which can reach several percent). Given the uncertainty of my listening test results, I would rather have you judge phono stages on topological considerations and measured performance. If you are still concerned about a preamp's sound quality, have your dealer arrange an in-home trial period. If your dealer will not allow an in-home trial you need another dealer. This service is one reason you are allowing your high-end dealer a 40-percent margin!

Line Stage Design

Table 1 shows the principal design elements of the line stages in this survey. I have also included kit designs by Borbely Audio and New England Analog (the latter run by William Snyder, who was the former design manager at Krell), the William Chater design (which appeared in the One/90, viz. February 1990, issue of *Audio Amateur)*, and the Deane Jensen JE-990 discrete op-amp (see Jensen 1980 in the references). The contents of the table may not be totally clear to those of you who are not familiar with analog amplifier design at the transistor level. But you do not need to understand all the columns to understand the points I want to make. If you are interested in this topic, I recommend the text *The Art of Electronics* by Horowitz and Hill. All you need to read this book is a background in high-school physics. *The Class A Design Manual* by William Snyder (published by New England Analog) is also a good introduction to the subject. This book is restricted to the design of high-end audio electronics. From this book you will find out more about class A design than Krell would like you to know. It is written at a higher level than *The Art of Electronics* and it is not nearly as pedagogical.

It requires no electronics knowledge to see from the table that no consensus exists on how to design a line stage. If the designers' goal was to produce a low-distortion, low-noise amplifier with low output impedance, we would not expect a consensus because many different topologies can be used to achieve these results. But some manufacturers also claim that a design goal for their line stages was "better sound." Such manufacturers claim that designing for good measured results is not adequate to achieve good sound quality. The designer will claim that some undiscovered X factor—that is, undiscovered except to the designer—must be considered to achieve good sound quality. But if a particular topology resulted in "better sound," one would expect many manufacturers, in time, to discover this topology. We would ultimately see a convergence to this single topology. But no convergence can be seen in Table 1. To me this indicates that no "X factor" exists.

I believe designers would be better off if they attempted to optimize their designs for best measured performance. A good example of this approach can be seen in an article by Erno Borbely in the August 1990 (Three/90) issue of *The Audio Amateur.* It could be argued that a 5534 op-amp has "good enough" numbers to render it inaudible. This may be true. But a similar argument would hold that a Ford Escort is "good enough" to get you from point A to point B. The preamps reviewed below are high-end luxury products, and "good enough" is not the standard by which they should be judged.

If you are familiar with audio amplifier design, Table 1 should be self-explanatory. One point I should make is in regard to open-loop gain. An amplifier with active loads in the first and second stages will have high open-loop gain at DC. An amplifier with resistive loads in the first and second stages will have much lower open-loop gain at DC. If a resistor is used only in the first stage, then the gain is between the two extremes. Detailed discussion of the line stages can be found in the individual test reports below.

RIAA Phono Equalizer Design

The RIAA phono equalization curve requires first-order poles at 50 Hz and 2120 Hz, which roll off the response at a rate of 6 dB/octave. A 500 Hz zero is also intro-

Table 1: Comparison of Significant Preamplifier Topologies

	<…LINE AMPLIFIER™								
	<—*Differential*			*Pair*—>		*Follower Stage*	<—*Second Gain Stage*—		
MODEL	*Active Element*	*Complementary Symmetry*	*Biased by Current Source*	*Cascode Stage*	*Load*		*Active Element*	*Cascode Stage*	*Load*
Acurus L10 (line) $599.00	Bipolar	Yes	No	No	Resistor	No	Bipolar	No	Active
Acurus P10 (phono) $395.00	NA	NA	NA	NA	NA	NA	NA	NA	**NA**
Adcom GFP-565 $799.95	NA(IC)	NA(IC)	NA(IC)	NA (IC)	NA (IC)	NA (IC)	NA(IC)	NA(IC)	**NA(IC)**
Aragon 24K $1500.00	Bipolar	No	No	No	Resistor	No	Bipolar	No	**Active**
B&K PRO10-MC $698.00	JFET	No	Yes	Yes	Resistor	No	Bipolar	No	Active
Borbely Audio[1] (partial kit form only)	JFET[2]	Yes	NA	Yes	Resistor	No	Bipolar	Yes	Active
Boulder "Ultimate" $5299.00	[JE-990-based]	[JE-990-based]	[JE-990-based]	[JE-990-based]	[JE-990-based]	[JE-990-based]	[JE-990-based]	[JE-990-based]	[JE-990-based]
Bryston .5B $795.00	Bipolar	Yes	No	No	Resistor	No	Bipolar	No	Active
Chater (partial kit form only)	JFET	Yes	Yes	Yes	Resistor	No	MOSFET	Yes	Active
Citation 21 $629.00	Bipolar	Yes	No	No	Resistor	Yes	Bipolar	No	Resistor
Coda 01 $2500.00	JFET	No	Yes	No	Resistor	No	MOSFET	Yes	Resistor
JE-990 (basic discrete op-amp)	Bipolar	No	Yes	No	Active	Yes	Bipolar	No	Active
New England Audio (partial kit form only)	Bipolar	Yes	Yes	No	Resistor	No	Bipolar	No	Active
PS Audio 5.5 $1195.00 (discontinued)	JFET	No	Yes	No	Resistor	No	Bipolar	No	Resistor
Jeff Rowland "Coherence One" $4600.00	JFET	Yes	Yes	Yes (Folded)	Resistor	No	NA	NA	NA
Sumo Athena II $828.00	Bipolar	Yes	Yes	No	Resistor	No	Bipolar	No	Resistor
Tandberg TCA-3018A $2299.00	Bipolar[5]	Yes	NA	No	Active	No	Bipolar	Yes	Resistor
Threshold FET ten/e $5700.00 (hl + pc)	JFET	No	Yes	No	Active	No	MOSFET	Yes	Active

[1]Dolan PM1 has similar phono amplifier
[2]Common-source input stage
[3]Output stage not in feedback loop
[4]Bipolar predriver stage
[5]Input connected to emitter follower which drives common-emitter input stage
[6]No global NFB used; Hawksford distortion-correction circuit used at output stage

NA = not applicable

(Not necessarily restricted to, nor including all, preamplifiers reviewed in this issue.)

----------> Push-Pull	Output Stage	<——Coupling Capacitors——> C_1	C_2	C_3	Total No. of Transistors	Power Supply Voltage	<——PHONO AMPLIFIER…………> 50 Hz Pole	500 Hz Zero	2122 Hz Pole	Notes
Yes	JFET	No	No	Film	8	±18	NA	NA	NA	
NA	NA	NA	NA	NA	NA	NA	See A	See A	See A	See E
NA (IC)	NA (IC)	No	No	No/Film	NA (IC)	±18	See A	See A	See A	See C/D/F
Yes	MOSFET	No	Electro-lytic	Film	8	±24	See B	See B	See B	See E Stage 2 is unity-gain
Yes	Bipolar	DC servo	DC servo	DC servo	13 + 1 IC	±30	See A	See A	See A	See E
Yes	MOSFET	DC servo	DC servo	DC servo	10 + 3 ICs	±24	Active Stage 2	Active Stage 2	Passive Aft. stage 1	Stage 1 for gain only
[JE-990-based]	[JE-990-based]	DC servo	DC servo	DC servo	?	±25	See B	See B	See B	SeeC/D Stage 2 for gain only
Yes	Bipolar	Film	Electro-lytic	Film	10	±24	Active Stage 1	Active Stage 1	Active Stage 2	Stage 2 is inverting
Yes	MOSFET	DC servo	DC servo	DC servo	16 + 1 IC	±25	Active Stage 1	Active Stage 1	Passive Aft. stage 1	Stage 2 for gain only
Yes	None	Electro-lytic	Electro-lytic	Electro-lytic	8	±23	See B	See B	See B	See C/D/F Stage 2 is unity-gain
Yes	Bipolar[3]	No	No	No	12	±30	Active Stage 2	Active Stage 2	Active Stage 1	SeeC/D
No	Bipolar	NA	NA	NA	9	Varies	NA	NA	NA	
Yes	MOSFET w/predr.[4]	DC servo	DC servo	DC servo	16	±25	Passive Aft. stage 1	Passive Aft. stage 1	Passive Aft. stage 2	3-stage design
No	MOSFET	Film	No	'Lytic + film byp.	10	±30	Passive Aft. stage 1	Passive Aft. stage 1	Passive Aft. stage 1	Active stages for gain only
NA	Bipolar	DC servo	DC servo	DC servo	?	±20	Passive Aft. stage 2	Passive Aft. stage 2	G_m cell, no global FB Stage 1	3-stage design (stg. 2 buffers 1)
Yes	Bipolar	Film	Electro-lytic	NP 'lytic +film byp.	10	±35	See A	See A	See A	See C/D/F
Yes	Bipolar[4-6]	No	No	Film	16	±22	Active: G_m cell, no global FB Stage 1	Active: G_m cell, no global FB Stage 1	RCntwk in G_m cell Stage 1	Stage 2 buffers stage 1 & adds gain
No	Bipolar	DC servo	DC servo	DC servo	11 + 1 IC	±18	See A	See A	See A	See C/D

A. All equalization is performed in one noninverting active stage.
B. All equalization is performed in the first active stage.
C. An extra zero is added to reduce the reactive load of the network on the stage.
D. The added zero is canceled by a passive network at some point in the circuit.
E. A zero at approximately 130 kHz is not canceled.
F. The output impedance of the phono stage is high.

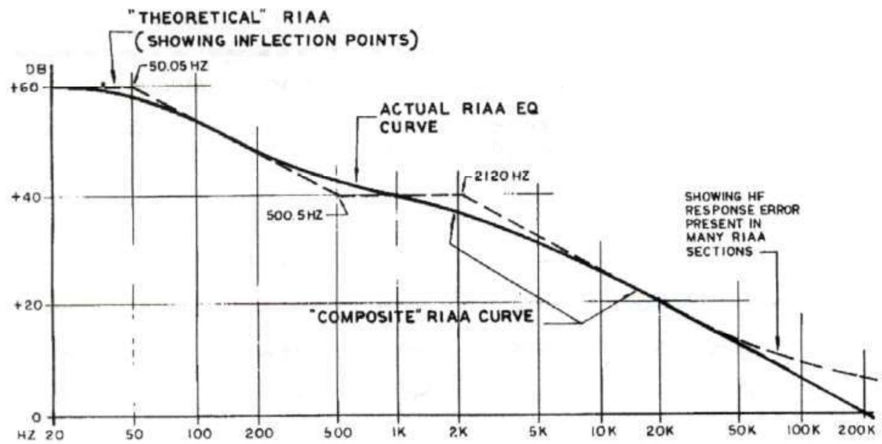

Figure 1: The finer points of the RIAA phono equalization curve (courtesy of Bryston Ltd).

duced into the transfer response. For typical "moving magnet" cartridges, below 50 Hz a gain of 60 dB is required. At 1 kHz this gain drops to 40 dB, and at 20 kHz the gain is reduced to 20 dB. In theory, the gain should continue to decline at a 6 dB/octave rate at ultrasonic frequencies. The complete RIAA de-emphasis curve is shown in Figure 1. Moving-coil cartridges require an additional overall gain of 20 dB beyond the figures given above. Designing an amplifier which is required to vary in gain by 40 dB (a ratio of 100 to 1) and to have a 60-dB low-frequency gain is not a trivial task. A number of very different design approaches can be used to solve the problems encountered in the design of a phono preamp.

The first problem faced by a designer is that he must have an amplifier which can drive large capacitive loads at high frequencies because the RIAA de-emphasis network becomes purely reactive at high frequencies. To reduce thermal noise from the resistors in the RIAA feedback network, the resistors should be made as low in ohmic value as possible. To keep the time constants in the network from changing, the values of the capacitors must be increased if the values of the resistors are decreased. So large capacitor values are required in the RIAA network if the phono stage is to have low noise.

A second potential problem with a phono stage is that its input impedance may not be constant. A variation in input impedance can lead to frequency-response errors, since the output of a phono cartridge can be very sensitive to loading. This effect can be minimized by designing the amplifier for high open-loop differential input impedance and high open-loop gain.

A third problem in designing a phono stage is maintaining the 6-dB/octave rolloff rate well into the ultrasonic region. To achieve this goal a noninverting amplifier stage cannot be used because this stage cannot be designed to have a gain less than one (0 dB) at any frequency. As a consequence, an undesired zero will be added to the transfer response at approximately 130 kHz, flattening it out above that frequency. Some designers argue that this zero is high enough in frequency to be ignored. But Chris Russell of Bryston argues that the zero can cause increased ultrasonic signals at the output of the preamp and that a power amp may have difficulty dealing with these ultrasonic signals. An inverting amplifier does allow signals to be attenuated to an arbitrary value but it is not appropriate for use in a single-stage phono equalizer. With an inverting amplifier, a 47-kilohm resistor would have to be connected from the input to the op-amp summing junction to achieve the required 47-kilohm input impedance. This resistor is in series with the input signal and its self-noise would lead to a noisier amplifier. RIAA network component values for the feedback loop would also become impractical under these conditions.

Often an additional resistor is placed in series with the RIAA feedback network to prevent the network from becoming totally reactive at high frequencies. This improves amplifier stability but at a cost. The parasitic zero now moves down to a frequency that cannot be ignored. Adcom, for example, uses a 500-ohm resistor in series with the RIAA network. The undesirable zero in the transfer response moves to 70 kHz when the resistor is added. In the Adcom this zero is canceled by a passive pole formed by an RC network at the output of the phono stage. This elegant solution is also used by Citation and Sumo. Because it has a relatively high output impedance caused by the passive RC network, it is very important that the output of this type of phono stage not be loaded.

An additional stability problem can occur because the RIAA equalizer requires a closed-loop gain variation of 100 to 1 within the audio band in addition to a low-frequency gain of over 60 dB. To insure low distortion at low frequencies, and to insure that the RIAA curve is accurately traced, the amplifier must have a gain 30 to 40 dB higher than its closed-loop gain. So, at low frequencies, we need an amplifier with more than 90 dB of open-loop gain. At higher frequencies we do not need this much gain. In fact, we do *not* want it because excess loop gain degrades stability. One solution to this problem is to contour the open-loop gain of the amplifier to match the closed-loop gain. The return-loop gain then becomes constant. Aragon and Citation create exactly this situation by removing the output stage of the amplifier and connecting the RIAA stage directly to the second gain stage. With increasing frequency, the RIAA network loads the amplifier and the open-loop gain of the amplifier decreases. But problems exist with this solution. An additional buffer stage is required to prevent the output load from affecting the phono stage. In the Aragon and the Citation this buffer stage is only a two-transistor circuit, which is marginally adequate. A second problem is that the RIAA network must not load the amplifier excessively, otherwise adequate gain will not be available at low frequencies. This limits the noise performance that can be achieved.

The Sumo phono stage is a very innovative design which eliminates this compromise. In the Sumo a traditional amplifier stage with an output stage is used. Sumo then adds a "dummy" network from the second gain stage to ground. This network shapes the open-loop gain of the amplifier. The "dummy" network is only responsible for setting the open-loop gain and it does not affect the noise performance of the phono stage. A buffer amplifier is not required since the output stage of the amplifier has not been removed.

Some manufacturers have determined that the problems of designing a RIAA equalizer can be more easily surmounted using two stages of amplification and dividing the required gain between them. It is much easier to design a stable low-distortion amplifier with high input impedance if the closed-loop gain requirements are reduced. It is also possible to build the phono stage to handle moving-coil cartridges directly without a pre-preamplifier because two stages are available to realize the required 80 dB gain. In a two-stage design the RIAA equalization also can be split between the two gain stages. This increases the degrees of freedom in designing the phono stage and can potentially lead to better performance. The loading effects of the RIAA network are reduced as is the sensitivity of pole position to component variation. For example, Bryston implements the 2120 Hz pole in the second stage of the preamplifier. Since this stage is driven by the low output impedance of the first stage, an inverting amplifier can be used without a performance penalty. No parasitic zero occurs in this implementation. The downside of a two-stage amplifier is that there is a second active stage in the signal path. If improperly designed, this could result in added noise and distortion. In any case it adds complexity and cost.

A two-stage design also allows all or part of the RIAA network to be implemented passively, with the passive de-emphasis network placed between the active gain stages. The second amplifier stage buffers the passive network from any loading effect at the output of the amplifier. The choice of how the gain is partitioned between each stage, the pole-zero assignments of each amplifier, and the method to realize the poles and zeros all pit the final noise level against the overload characteristics of the amplifier. If these parameters are improperly chosen, the signal swings at the output of the first gain stage can become very large at high frequencies. This problem is most likely to occur when passive RIAA networks are used. Table 1 includes the pole-zero assignments and active/passive network partitions for a number of preamps. No optimal solution to this assignment problem has been identified; manufacturers are using almost all the possible permutations. Some manufactures would claim that the partitioning decisions are made, in part, on sound quality considerations. Again the fact that no convergence to a single method is observed indicates that no magic "X factor" has been discovered.

Measurements

This section is a brief discussion of the testing methodology we used. I am not going to use a lot of space discussing the fundamentals of the topic, since an excellent discussion of those fundamentals can be found in an article by Erno Borbely, which appeared in *The Audio Amateur* [Borbely 1989]. Briefly, the performance of an audio amplifier can be characterized in three distinct categories: (1) the linear errors of the amplifier, (2) the nonlinear errors of the amplifier, and (3) the noise added by the amplifier.

Linear errors are characterized by changes in frequency response. We measured the change in the magnitude of the frequency response only in the region of human hearing (20 Hz to 20 kHz). I see no reason to penalize a designer who chooses to improve RF immunity at the expense of a rolloff in the ultrasonic region. Similarly, the designer who chooses to use small film capacitors for DC blocking instead of large electrolytics should not be penalized because his amplifier rolls off in the subsonic region. Tweaks and cultists, please note that small frequency response errors are audible and that these errors account for most audible differences in electronics. We found no frequency errors significant enough to report on in any of the preamps' line stages. Some phono stages did exhibit errors large enough to be audible.

We did not measure the phase portion of the frequency response of the amplifiers. If an amplifier has no transmission zeros in right half plane in the region of human hearing—this holds for all units reviewed below—then it will not exhibit any significant phase error. Tweaks claim that even small phase deviations may be audible, but no studies in scholarly journals support them. The tweaks ignore the fact that microphones, cartridges, and speakers all have significantly more phase errors then any electronic component. It is also interesting to note that tweaks never worry about the principal source of phase errors in a preamplifier. These phase errors are caused by small errors in the RIAA equalizer of the phono stages.

Tweaks also complain that steady-state sine wave testing does not show all errors that will occur in a music signal because the music is not deterministic. With respect to linear errors, it can be shown using spectral analysis [Papoulis 1984] that nondeterministic signals are affected no differently by linear errors than deterministic signals. In other words, if an amplifier has a flat frequency response in the range of human hearing and has no nonlinear errors, then it will pass a music signal unchanged.

Errors due to nonlinearities in the amplifier's transfer response can be broken down into two parts: static and dynamic. A nonlinear transfer function can be characterized using a power series expansion. Terms of the expansion that are frequency-independent result in static distortion. Fre-

Capacitors in the Signal Path

Audiophiles have recently become concerned about capacitors in the signal path of active electronics [Jung and Marsh 1980]. They believe that the capacitors can change the sound of electronics. This concern probable occurred when it was observed that the presence of an electrolytic capacitor could be heard in loudspeaker crossover networks. But in that application a capacitor is required to swing large voltages into low-impedance loads (the loudspeaker drivers). Under those conditions measurable distortion due to the electrolytic capacitor may occur. In a preamp, the voltage swings are much smaller and the loads larger by at least three orders of magnitude. Under these conditions frequency nonlinearities in the range of human hearing should not be measurable, but some inband time-domain effects have been reported [Jung and Curl 1985].

Figure 2 shows the places where capacitors are used in an active amplifier stage. Capacitor C_1 rejects any DC present on the incoming signal line. In addition it blocks any DC bias current required by the gain stage from being sourced from the signal driving the gain stage. The bias current is supplied through R_1 instead. In a phono stage any DC current flowing through the cartridge could cause a misalignment of the cartridge cantilever. In a line stage the DC current flowing through the balance and volume controls could lead to noise when the controls are operated. If the presence of DC from the signal source is not a concern, then C_1 can be eliminated if a FET input stage is used instead of a bipolar stage. The FET does not require a DC bias current. C_3 blocks any DC offset present at the output of the gain stage from passing to the next stage.

If the gain of the amplifier is large, then the DC level at the output of the gain stage be high. While C_3 prevents this DC from passing to the next stage, a problem remains. A large DC offset will result in asymmetrical clipping at the output of the gain stage. This will reduce the dynamic range of the amplifier. Placing C_2 in the feedback loop solves this problem. To see why, recall that the gain of a noninverting amplifier is given as:

$$A_v = R_3/Z+1$$

where Z is the impedance of the series combination of R_2 and C_2. If the R_2C_2 time constant is set well below 20 Hz, then the amplifier will have a gain in the range of human hearing given by the formula:

$$A_v = R_3/R_2+1$$

but, at DC, Z and $A_v = 1$. Thus the DC offset is not amplified. Often the use of C_2 in conjunction with an amplifier with a low offset voltage results in a DC level of less than 10 mV at the output of the amplifier. Under these conditions C_3 may be eliminated. Adcom uses this technique in the phono stage of its GFP-565. Adcom's promotional literature clams that the preamplifier is direct-coupled but this is clearly not true.

The mechanism by which C_1 and C_3 can cause distortion in the signal path is easy to see. Any nonlinearity from the capacitor C_1 or C_3 appears directly in series with the input or output signal, respectively. The effect of C_2 is harder to see. But C_2 is in the feedback loop of the amplifier and any nonlinearity in C_2 will distort the output signal. Here is how this distortion occurs. If the gain of the amplifier is infinite, then the voltage across the summing junction is 0, and V_{in} appears across R_2 assuming that $Z_{C2} \ll R_2$. This last inequality will hold in the audio band for any correctly designed amplifier. If the potential V_{in} is across R_2, then a current $I_2 = V_{in}/R_2$ must flow through R_2. Since no current can flow in the summing junction, the current must also flow in R_3. Since R_3 is connected between V_{in} (recall there is no drop across the summing junction) and V_{out}, we have:

$$(V_{out} - V_{in}) = R_3 I_2 = (R_3/R_2)V_{in}$$

Performing simple algebra yields the standard gain relationship for a noninverting op-amp:

$$V_{out} = V_{in}(1 + R_3/R_2)$$

Now consider a nonlinearity in C_2 which results in a voltage drop across C_2. This will change the voltage across R_2 and thus the value of I_2 and in turn V_{out}—clearly a distortion mechanism.

The impedance of R_3 and R_2 may be significantly smaller than R_1 and R_4. This results in the requirement that C_2 must be significantly larger than C_1 or C_3 to prevent frequency-response errors at 20 Hz. For example, the Adcom uses a 2200 µF electrolytic capacitor for C_2 in the phono stage of the GFP-565. C_1 and C_3 are often small enough that they can be film capacitors, but C_2 must almost always be an electrolytic (see Table 1). An example of this can be found in the Bryston .5B preamp. In the Bryston, C_1 and C_3 are polyester capacitors and C_2 is an electrolytic capacitor. The smaller impedance of R_2 and R_3 also results in larger displacement currents flowing through C_2 in comparison with C_1 or C_3. Thus it can be seen that C_2 is the most likely source of any distortion at the output of an amplifier that can be attributed to capacitor nonlinearities. This is because C_2 is physically larger than C_1 or C_3, and in addition the displacement currents which flow through C_2 are higher than the currents which flow through C_1 or C_3.

Curl, Jung, and Marsh among others recommend that large electrolytics be bypassed with a smaller film capacitor. At higher frequencies, where the electrolytic capacitor becomes inductive, the film bypass capacitor carries the signal and, in theory, minimizes the effect of the nonlinearities of the electrolytic. Chris Russell of Bryston and Marty Zanfino of Harman Kardon have both independently pointed out a problem with this approach and they do not bypass electrolytics. They found that the parasitic inductor in the electrolytic and the film capacitor can form a resonant circuit. They report that at some frequencies the combined capacitors may perform more poorly than an unbypassed electrolytic capacitor.

A properly designed DC servo can eliminate the need for C_1, C_2, and C_3 but it adds complexity, and a failure of the servo network can be catastrophic. When the servo fails, DC levels of 10 volts or more can appear at the output of a preamp. Ideally a DC sensor should be placed at the preamp's output to protect power amps or speakers, but this is rarely done.

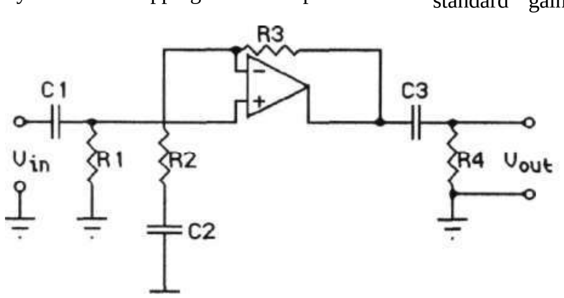

Figure 2: Capacitors in an active amplifier stage.

quency-dependent terms are grouped under the category of dynamic distortion. The principal source of static nonlinearity is the nonlinear transfer function of the active devices. If a sine wave is applied to an amplifier having a nonlinearity, its output will contain additional sine waves with frequencies at integer multiples of the fundamental. Total harmonic distortion measures the magnitude of the power of these signals relative to the power of the fundamental. The larger the nonlinearity, the larger the value of the THD. Thus we can use the THD test alone to characterize the static nonlinearities of the amplifier. Tweaks please note that nondeterministic music signals act no differently in the presence of static nonlinearity than deterministic sine waves. Thus, if an amplifier shows very low values of THD when a low-frequency sine wave input is used, then static nonlinearities will have no audible effect.

Dynamic nonlinearity has a multitude of origins in modern electronic amplifiers. One source, in the case of amplifiers with high open-loop gain at DC, is the declining open-loop gain at high frequencies, which in turn reduces the return-loop gain. The open-loop distortion is reduced by a factor proportional to the return-loop gain. Distortion, thus, increases as the input frequency increases. An additional effect of reduced open-loop gain is that the signal level at the summing junction of the op-amp increases as frequency rises. With an increased signal level at the summing junction, the distortion caused by the first stage of the op-amp increases [Jung 1979]. The combination of the two effects described above can result in distortion which rises at 60 dB per decade [Wurcer 1992]. Another source is that sufficient current may not be available to charge capacitive loads or capacitive internal nodes. Slew-rate limiting is the most extreme example of this effect. The nonlinear parasitic capacitors in active elements can give rise to dynamic distortions [Cherry 1983]. Marginally stable circuits can also cause dynamic distortion [Borbely 1989].

Reducing dynamic distortion is often a difficult problem. For example, increasing the current available to charge the capacitor that sets the dominant pole of the amplifier (the compensation capacitor) also increases the open-loop gain of the amplifier. With the open-loop gain increased, the size of the compensation capacitor must be increased to insure stability. For an undegenerated bipolar amplifier the compensation capacitor must be increased by the same magnitude as the current, and nothing is gained [Gray 1984].

Another example of the engineering trade-offs in reducing dynamic distortion involves the setting of feedback levels. Consider an attempt to increase the feedback level in order to reduce distortion. To increase the level of feedback, the gain of the amplifier must be increased, but at high frequencies the gain of the amplifier is predetermined by stability conditions. This predetermined level of gain is related to the gain-bandwidth product of the amplifier. If the amplifier has been properly designed, its gain-bandwidth product will be limited by the active devices used. Thus, without changing the active devices, it is not possible to increase the level of feedback. Active devices with exceptional high-frequency performance will have other suboptimal performance characteristics which usually prevent their use in a given amplifier design. The engineering trade-offs described above typically limit the return-loop gain of a phono or line stage in a preamplifier to between 10 and 100 at 20 kHz. These return-loop gain figures represent low to moderate feedback levels. I hope this discussion puts to rest any untutored talk about the use of high feedback as a simple way to reduce distortion.

A simple method to characterize the worst-case dynamic nonlinearity is to measure the THD of the amplifier at 20 kHz [Jung 1979], This signal has the highest rate of change of any inband signal. The disadvantage of this approach is that the distortion products are out of band. A second disadvantage is that the upper band limit of the amplifier under test attenuates higher-order harmonics. Multiple-tone IM tests resolve these problems [Borbely 1989] but they do not operate at the maximum input rate of change, may characterize only even- or odd-order nonlinearities, and are more difficult to implement. We did not measure phase intermodulation distortion (PIMD) [Otala 1980] because it has been shown that the effect is small in magnitude [Cherry 1983] and the occurrence of PIMD will always be accompanied by significant amounts of intermodulation and harmonic distortion [Cordell 1983].

Figure 3 shows a typical THD vs. output level curve set. Measurements are made at three frequencies: 20 Hz, 1 kHz, and 20 kHz. The distortion does not increase at low signal levels as implied by the curve. The Audio Precision distortion analyzer routine we used does not distinguish between noise and distortion. The total power of all signals located at frequencies other than the fundamental frequency is divided by the power of the signal. At low signal levels the noise dominates the measurement. Since the noise power is a constant, the ratio increases as the signal power decreases. Because of the noise it is not possible to see that the distortion levels are actually getting lower as the signal level decreases. Noise dominates line-stage measurements much more than measurements of the analog sections of CD players because line-stage amplifiers have voltage gain and CD player stages do not. The input-referred noise of the amplifier is increased by the voltage gain of the amplifier. Often a designer must trade off amplifier noise performance to linearize the front stage of the amplifier. The linearization process reduces dynamic linearity errors. If the designer makes this trade-off, noise at the output of the line stage is further increased. The curve shown in Figure 3 is almost totally dominated by noise. Only at 15 volts out does the ratio of noise to signal level become low enough for the distortion to become visible. The very rapid rise in distortion takes place when the signal clips because the signal comes to close to the supply rails. For the 20 Hz and 1 kHz curves shown in Figure 3 the residual distortion value is so low that it is not actually identified in the curve. The amplifier clips before the signal-to-noise ratio becomes large enough to expose the distortion.

From Figure 3 it can be seen that the 20 kHz curve de-

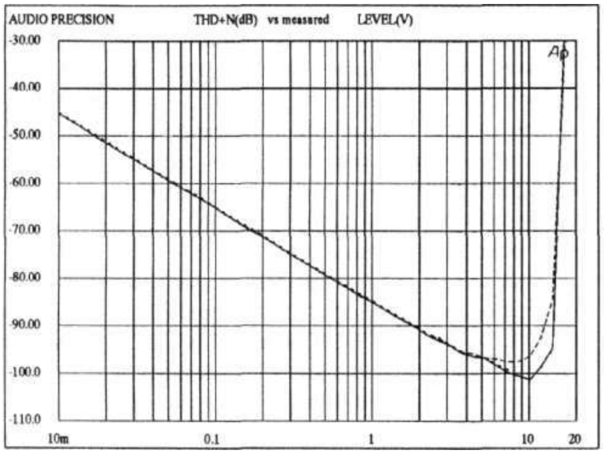

Figure 3: THD plus noise versus output level in an excellent line stage at 20 Hz (short dashes), 1 kHz (solid line), **and 20 kHz (long dashes).**

viates from the other curves at a little over 5 V rms. This is an example of dynamic distortion. For this line stage the deviation is small and occurs at such a high signal level as to be insignificant, since most power amps will produce maximum power when the preamp output signal exceeds 2 V rms. With phono stages amplifying high-gain cartridges it is possible for significant output excursions to occur. The signal may swing 5 V rms or more. Under these conditions the volume control is adjusted to attenuate the signal into the power amp. Clipping levels of the phono stages are reported in the individual reports below to alert you to preamps which would have problems with high-gain cartridges.

In addition to measuring distortion with the outputs unloaded, we measured the distortion with the outputs loaded into 600 ohms. This allowed a better characterization of the distortion performance of the output stage. No modern single-ended power amplifier has an input impedance lower than 10 kilohms. As a safety factor, preamps with well-designed output stages should show almost no increases in distortion when the 600-ohm load is added.

All measurements given in this survey are of distortion plus noise relative to the signal level used for a given test. The measured distortion results are reported in decibels. This is far more readable than an linear representation of distortion. A 20 dB difference in distortion represents a decade change in the magnitude of the distortion. If you want to convert the dB figures back into the traditional percent figures, the following should help:

-20 dB = 10%	-80 dB = 0.01%
-40dB = 1%	-100 dB = 0.001%
-60dB= 0.1%	-120 dB = 0.0001 %

It may be possible for humans to distinguish distortion which is one part in a hundred (-40 dB). I am not aware of any controlled experiments which show that distortion at the level of one part in a thousand (-60 dB) is audible. The best preamps in this survey produce distortion at the level of 20 parts in a million (-94 dB) at signal levels well in excess of what is required to clip any power amp. Music signals which have much lower rates of change than a 20 kHz sine wave would be expected to exhibit even lower values of dynamic distortion. Also note that music signals are typically much smaller in amplitude than the test signal levels used here. Dynamic distortion is lower for smaller signal levels, and distortion will be far below the noise level when a preamplifer is reproducing music

We did not perform any square wave tests. Nonband-limited square waves can indicate sub- or ultrasonic deviations in frequency response. As explained above, these deviations have no audible consequence. Square wave testing is often used for identifying dynamic linearity problems (see, for example, Figure 4 of Bascom King's review of the Motif MC8 preamp in the July 1989 issue of *Audio).* This information is often very useful to the designer or service technician attempting to find the cause of the nonlinearity. Square wave testing is not required, however, to evaluate a product because, if an amplifier shows dynamic linearity problems in square wave testing, it will also show the problem on sine-wave-based THD tests (from Figure 1 in the Bascom King review of the Motif we see distortion levels of 2% at 20 kHz)- In general, the THD test will show small dynamic linearity effects (-60 dB and below) that cannot be seen on square waves.

Stereo separation was measured using a standard program developed by Audio Precision. The program is written so that only crosstalk of the fundamental test frequency is reported. Noise of the channel under test is removed by setting the tracking bandpass filter, built into the Audio Precision, at the fundamental frequency of the test signal.

Phono stage input impedance characteristics were evaluated using a novel method proposed by Bob Carver. The input stage was placed in one leg of a balanced bridge. Another leg of the bridge was connected to a variable resistor and variable capacitor. The bridge is driven by a sine wave source which can be varied in frequency and amplitude. If the bridge cannot be brought into balance by adjusting the variable resistor and variable capacitor, then the phono stage's input impedance cannot be modeled by a simple RC network. A complex input impedance may cause phono cartridge frequency errors. All the phono preamps below passed this test.

Phono noise was characterized using a method different from the IHF standard. The first difference is that we do not weight the noise. Weighting reduces the sensitivity of the noise test to power-supply hum and 1/f noise (which decreases 6 dB per octave) from active devices. The second difference is that we measured the noise at the output of the phono stage and then referred it back to the input. The voltage gain of the preamplifier at 1 kHz was used for this calculation. The ratio of the maximum voltage output of the cartridge at 1 kHz (this should be obtainable from your cartridge's manufacturer) to the input-referred noise of the preamp gives the signal-to-noise ratio of the complete phono system. You would want a minimum ratio of 2000 (i.e., 66 dB) for high-quality reproduction.

In summary, we are using traditional testing methodologies to evaluate the preamps below. No special tests have been developed because they are not required. The measured results for the line stages were so good, for all preamps discussed below, that we should expect no audible differences which can be distinguished between the units (except perhaps units with poor channel separation). Our own double-blind (ABX) testing yielded no positive results for the line stages, confirming the predictions based on the electrical measurements. Reports of audible differences in nonblind listening tests are in direct contradiction to the analytical and empirical results presented in this article. Such reported differences, like earlier astronomical observations reporting canals on Mars, appear to be in the mind of the observer and not present in the observed object.

Acurus L10

Mondial Designs Limited, 2 Elm Street, Ardsley, NY 10502. Acurus L 10 Line Preamplifier, $599.00. Tested sample on loan from manufacturer.

One look at the schematic of this preamp shows the fingerprints of William Snyder. Who is William Snyder? He used to be the design manager at Krell and is now a consultant to Mondial. So what we have here is basically a Krell topology for $600. Both the Krell and the Acurus have fully discrete power-supply regulators with no global feedback and both units share a complementary, fully discrete line-stage topology. You also get Krell build quality. The PC board is double-sided with plated-through holes. Auto-inserted components are wave-soldered to the very spacious, well-laid-out circuit board. The volume and balance controls are laser-trimmed, sealed Nobel units. The output capacitor is a very expensive 10 µF polypropylene unit. The sheet-metal work is excellent.

This is a full-function line-level preamplifier with six inputs and a separate record function selector. No Krell preamp gives this much flexibility. The two tape-monitor outputs are paralleled and driven from a single wafer on the record function selector. This economy design feature allows a potentially destructive oscillation to occur if the record and input selector are both set to Tape 1 (or both to Tape 2). This is the same problem we found on the $2500 Coda preamp (see Issue No. 16). Clearly the compromise is more understandable in a $600 preamp, but I would not unconditionally recommend this preamp for a user with multiple tape recorders. The inputs are direct-coupled to the line stage (C_1 is missing). If this is done incorrectly, clicks or other noises are emitted from the output of the unit as the controls are operated, but this problem did not occur with the Acurus.

Ergonomics on this preamp are superb. The large front panel is equipped with five evenly spaced and well-marked knobs. The full-sized (2 centimeters deep, 3 centimeters in diameter) metal knobs convey a sense of luxury you expect only in a premium-priced unit. The 2-inch-high Krell design with its thin control knobs is not as convenient to use as the Acurus design. Nothing plastic is found on the front panel of the Acurus. The controls feel very solid. One reason for this is the front-panel mounting of the input selector and record selector switches. Often these controls are mounted at the rear of a preamplifier and connected to the front-panel knobs through long, rotating shafts. By placing the selector switches near the input jacks, single-sided PC board layout is significantly simplified. The double-sided board of the Acurus does not require this compromise. Left-channel signals are run on one side of the board, and right-channel signals are run on the other side. Careful use of ground planes prevents crosstalk between channels and unselected inputs. This care was reflected in the superb measured crosstalk. Even in the less good channel it stayed below -90 dB up to 1 kHz and then rose at a 6 dB per octave rate to -71 dB at 20 kHz. For comparison, identical measurements made by *Stereophile* on preamps costing up to an order of magnitude more than the L10 are not this good at 20 kHz. For example, *Stereophile* measured a separation figure of -46 dB at 20 kHz on the $2000.00 Krell KSL, and the $4500.00 Krell KBL measured -60 dB at the same frequency. The only preamp measured by *Stereophile* that performs better than the L10 is the Coda 01, which stays below -100dB across the complete audio band. *[Regrettably, we had no access to an Audio Precision System One when we tested the Coda 01 for our review in Issue No. 16, otherwise we would certainly have pointed out such an outstanding test result.—Ed]*

It is interesting to note that the L10 also had a flatter frequency response and lower THD-plus-noise figures (for 300 mV input drive) when compared with the measurements made by *Stereophile* (April 1992, page 248) of the Krell KBL line stage. (The KBL was designed after Mr. Snyder had left Krell—perhaps they should have tried harder to keep him.)

Some cost compromises are nevertheless evident in the Acurus L10. The transformer is a small, low-current unit; heat sinks are not used on the regulator pass transistors, and resistors rather than current sources are used to bias the differential pairs in the gain stage. Regulated power-supply rail voltage is comparatively low at ±18 V. In contrast, the Sumo has ±35 V supply rails. Each channel has a separate regulator pass transistor, but both channels share the same diode reference. The output stage of the gain stage uses small-signal JFET output transistors which are not run at very high bias currents. The full-blown William Snyder line-stage topology does not have these compromises (see Table 1). The bias currents are high enough to ensure that the output stage remains biased in class A when loaded by any modern power amplifier.

Measured distortion performance of the L10 was excellent but it was bettered by B&K and Sumo. With the outputs unloaded, the 1 kHz THD plus noise reached a minimum of -92 dB at 9 V rms. The 20 kHz distortion reached a minimum of -86 dB at 5 V rms. Adding a 600-ohm load to the output did not significantly affect these results. This puts my concern about the JFET output transistors to rest.

All is not perfect with this unit, however. Of greatest concern to me is the elimination of the output muting circuit. The unit does not emit a turn-off pulse but it does have a small turn-on pulse that can result in a large—but not system-destroying—pulse at the speaker terminals. The unsealed switches are only silver-plated, so you should not expect 20-year reliability from this unit.

In summary, I can see no difference between a Krell and the Acurus L10 that would affect reliability or sound quality. Overall I give a "best buy" recommendation to the Acurus L10 provided a phono stage is not required. Acurus is an acronym for "AcCURacy in the US," a well-chosen name. If we could just get GM to build cars with the same quality, value, and reliability as this preamp, then the USA would be Number One again.

Acurus P10

Mondial Designs Limited, 2 Elm Street, Ardsley, NY 10502. Acurus P10 Phono Preamplifier, $395.00. Tested sample on loan from manufacturer.

This unit is a separate phono amplifier designed to match the L10, but it can be used with any line-level-only preamp. This is a very expensive approach to adding a phono stage, since the metal chassis and power supply must be duplicated (and paid for, bringing the P10's price up to almost $400). Compare this with the $149 Sumo charges for an optional phono PC board which can be installed in its Athena II preamp and with the $45 price difference between the Bryston .4b and the Bryston .5B, which differ only by the fact that the .5B has a phono stage. Additional design problems with this approach occur because the phono stage must be designed to drive cables and the unknown load of the line. Since the Acurus L10 has no tape-monitor buffers, tape-recorder input loads must also be driven by the phono stage.

The construction quality and power supplies of the P10 are the same as in the L10. As the ultimate test of P10's build quality, I showed it to Anne Morea, a military certified quality inspector whose work I respect highly. Anne is feared because she has shut down military production lines. She found a number of minor problems, some which were related to minor violations of military construction specs that this unit is not required to meet. The most serious problem she found was some potentially cold-soldered joints on a reworked diode. Overall she was impressed enough to declare that this unit came as close as any commercial equipment she has examined to meeting military specifications. Believe me, from her this is very high praise. One construction highlight of this unit is the use of gold-plated shorting bars for setting phono input loading and phono stage gain. This is much more reliable than using switches that are not gold-plated.

The P10 amplifier stage is almost identical to the L10 amplifier stage, but because of the higher gain of the phono stage C_2 (see sidebar) must be added; an electrolytic bypassed with a film capacitor is used for C_2. There is no C_1. Acurus relies on matching of the *npn* and *pnp* base currents in the symmetrical differential stage to reduce input current flow to an acceptable level. Sumo does the same thing. I would be happier if C_1 were included to prevent even residual current flow in the phono cartridge. A bigger problem is the elimination of the output muting circuit. Turn-on and turn-off transients could cause damage to power amplifiers and speakers.

No moving-coil stage is included in this unit. Instead, the value of R_2 can be changed (see sidebar) to increase the amplifier's gain. In high-gain mode this unit has a closed-loop gain of 80 dB at 50 Hz. For low distortion and good RIAA tracking, we need an open-loop gain of 110 dB under these conditions. Measurement data below shows that the Acurus amplifier stage cannot achieve this number. Furthermore, the symmetrical differential pair used in the first gain stage is not a very low-noise topology. Additional problems occur because the RIAA network does not have an added resistor to reduce reactive loading of the network on the amplifier. Acurus could not use this technique because it requires an additional passive network at the output of the phono stage. This passive network would raise the output impedance of the phono stage to unacceptably high levels for driving cables plus a line stage of unknown characteristics. Without the resistor, the output must drive the series combination of a 1.2 nF capacitor and a 100-ohm resistor. I expected that this would drive the self-biased JFET output stage deep into class B mode, with an increase in 20 kHz distortion evident. This result was not seen in the measurements, perhaps because the overall distortion was relatively high.

Clearly this preamp should be used only at its lowest gain setting, where it is required to drive only a 1-kilohm load in series with the 1.2 nF capacitor. But even in this mode, we are left with a single-stage design that lacks the sophistication of the Bryston phono stage.

Measured performance was, not surprisingly given the above analysis, a mixed bag. RIAA equalization error was ±0.35 dB in the medium-gain mode. The response declines monotonically from 20 Hz to 20 kHz. It looks very much like the response of a Quad preamp tilt control. This is almost certainly audible. I can just imagine the reviews in the underground magazines—great pace and rhythm, the strings were silky smooth. The imbalance between the channels was remarkably small, since it never exceeded 0.025 dB. This indicates that component variation is not responsible for the error. It is instead the result of incorrectly calculated RIAA component values. The input-referred noise level in the worse channel at medium gain was 0.35 µV. This is a truly excellent result, especially in view of the fact that the symmetrical differential pair used in the first gain stage is not normally (see the Bryston and Sumo results below) a very low-noise topology.

The 1 kHz and 20 kHz THD plus noise reached broad minima of -76 dB from 6 V rms to 10 V rms. These are below-average results. The 20 Hz results were even less good, reaching a minimum of -68 dB at 5 V rms. As described above, the unit does not have a large enough return-loop gain at low frequencies at the medium-gain or high-gain setting. At the high-gain setting, 20 Hz THD never was lower than -54 dB. The curve had a broad minimum from 2 V rms to 6 V rms. In the high-gain mode the RIAA error curve developed a 0.4 dB rolloff starting at 100 Hz and continuing to 20 Hz. This is another indication that the return-loop gain is not adequate in the high-gain setting. One bright point of the high-gain setting was a very low input-referred noise value of 0.2 µV. The decrease in noise at the high-gain setting indicates that the equalization network, not the discrete op-amp, is the dominant source of noise. Phono overload at all gain settings is very good, since the output of the P10 clipped at 12 V rms at all frequencies.

What Acurus needs to do is to raise the price of the unit so that a moving-coil stage can be incorporated. In addition, a more advanced main phono stage should be used so that this unit can approach the state of the art. In fact, the phono stage of the Aragon (another brand of Mondial Designs) 24K preamp, with an improved output buffer, would be a good place to start. The 24K also has a good moving-coil head amp. Correctly done, this should add only $100 to the price of the phono stage. For the present, if you need a phono stage, choose a preamp in this survey that has one built in.

(This is our first negative reaction, ever, to a Mondial product. We would like to evaluate other Mondial products, including the Aragon power amps, which look very promising. Audio company personnel tend to have rather thin skin, and it will be interesting to see just how willingly Mondial will continue to lend us products after this review.)

Adcom GFP-565

Adcom, 11 Elkins Road, East Brunswick, NJ 08816. Model GFP-565 preamplifier, $799.95. Tested sample on loan from dealer.

"Why does this $800 preamp have cheap plastic knobs, and does this indicate that cost compromises have also been made inside?" That was my first thought when I examined the Adcom GFP-565. Unfortunately, the plastic knobs were not the only sign of cost cutting in this preamp. The first thing you notice when the unit is opened is the large PC board, which is manufactured and assembled by Rotel in Taiwan (according to our usually reliable sources). The board does not have plated-through holes because the top foil side of the board is used only as a ground plane. Topside component interconnections are made by numerous jumpers. Nine bus bars run across the center of the board to reduce the impedance of the power supply runs. These bus bars would have been unnecessary if Adcom had chosen to use a double-sided board with plated-through holes and thicker (2-ounce) copper traces. Such a PC board would allow low-impedance, efficiently routed power supply runs but it would cost almost twice as much as the board Adcom uses. Both B&K and Acurus use the more expensive double-sided board with plated-through holes.

The sheet metal work is typical of mass-produced consumer electronics; the unit is held together by sheet metal screws. The headphone jack is not well mounted to the front panel. The chassis and housings of Acurus, B&K, and Bryston units are built to a higher quality standard. The power transformer does not appear to be well mounted to the main cabinet. This contrasts sharply with the externally mounted toroid transformer used by B&K. The volume and balance controls used by Adcom are unsealed Alps pots which are less expensive and less reliable than the sealed, laser-trimmed Nobel pots used by B&K, Acurus, and Sumo. The function selector switches are unsealed linear switches. It is unclear whether these or any other switches in the Adcom unit are silver-plated. The front panel has a rotational to linear converter which drives the switches through a long unsealed band of metal. This arrangement could be a long-term reliability problem if the band of metal starts to bind or stick in its unsealed channel. On the positive side, the Adcom has separate switching for each tape recorder on the record function selector. This prevents the input and output of a tape recorder from being connected together and creating an oscillation that could damage a loudspeaker. The Acurus does not have this feature, and the B&K can drive only one tape recorder.

The power supply of this preamp is one area which did not show evidence of cost reduction. The transformer is a large, high-current design. Large 6800 µF capacitors filter the unregulated supply rails. The voltage regulator is a discrete design with global feedback. Regulator pass transistors are placed on generously sized heat sinks. A single voltage regulator drives both the left and right channels. The regulated supply voltage is held to 18 V because integrated circuits are used for voltage amplification in this design. Relay-based muting circuits prevent turn-on and turn-off transients from appearing at the preamp's output. This is one of the most important features that a preamp can have, since the transients can be large enough to damage loudspeakers. A muting circuit is absent from both the Acurus and the B&K.

Three line outputs are available. The only difference between two of them is that one is direct-coupled and the other has a bypass capacitor. The third output differs from the other two by the fact that it is connected before the tone-control defeat and highpass filter switches. The other two outputs are connected directly after these switches. There is no way you could "hear" these switches on Day One. I wonder if, by providing a bypass around the switches, Adcom is indicating it is worried about the reliability of the switches used in this unit.

All of the integrated op-amps have Adcom part numbers. These are not custom-made parts manufactured exclusively for Adcom but just custom-tested versions of standard integrated circuits manufactured by Linear Technology. Intelligent guessing can be used to identify the parts used by Adcom. The phono stage very likely uses the LT1028 op-amp or the similar LT1115, which has relaxed DC specifications. The LT1028 is an improved version of the popular NE5534 op-amp used in many mid-fi phono stages and CD-player outputs. The LT1028 has lower noise and distortion when compared with the NE5534 [Duncan 1990], but both of these op-amps have very small (0.036 V) slewing thresholds (called V_{th} and expressed as the slew rate, in V/µS, divided by the 2π multiple of the gain-bandwidth product, in MHz) [Jung 1986]. A small slewing threshold limits the maximum size of the input signal that can be presented to a gain stage if dynamic distortion is to be prevented. It can be argued that the small voltage output of a phono cartridge reduces the slewing threshold requirement of an op-amp when it is used in a phono stage. In addition, the high gain-bandwidth product of the LT1028 ensures a sufficient return-loop gain at 20 kHz to linearize the distortion from the input stage [Jung 1979].

Op-amps used in the line stage are harder to identify because many devices manufactured by Linear Technology could be used. The line stage of the Adcom probably uses the LT1056, which has a slewing threshold of 0.4 V. (Other potential candidates include the LT1022, which has a slewing threshold of 0.5 V, and the LT1122, which has a slewing threshold of 0.9 V. The LT1056 is less expensive than these last two chips. The LT1056, LT1022, and LT1122 all have higher noise and distortion in comparison with the LT1028.) There is a compromise between noise, distortion, and slewing threshold in IC designs because the lateral *pnp* transistors in a standard IC process are very slow, and the total silicon die area for the complete integrated circuit is limited if the device is to be economical. This compromise does not have to made in a discrete operational amplifier.

As is the case for almost all op-amps, the chips used by Adcom have class AB output stages. Class AB stages can exhibit crossover distortion as the stage "transitions" from class A to class B operation. Adcom works around this problem by connecting a current sink at the output of the IC. This forces the IC into class A operation. This is an effective solution but it increases the die temperature of the IC, which may result in long-term reliability problems. The discrete amplifiers used by Acurus, B&K, Bryston, and Sumo are designed, unlike IC op-amps, to run in the class A mode, and the work-around used by Adcom is not required.

A third problem with IC op-amps is limited current driving ability. Adcom overcomes this problem by using a buffer chip, which is almost certainly the LTI0I0. This IC uses an *npn* emitter follower for sourcing current. An *npn* common-emitter output transistor, in conjunction with an op-amp wired in a unity-gain configuration, is used for sinking current. This complex quasi-complementary circuit is required because the on-chip *pnp* devices in a standard IC process are not fast enough to be used as output devices. High-speed *pnp* devices are available on more advanced process technology using dielectric isolation. Harris Semiconductor, among others, has developed buffers using this process. These buffers have better performance than the LTI0I0 but are much more expensive. From the Linear Technology data sheet on the LTI0I0 we find that "the scheme is not perfect in that the rate of rise of the sink current is noticeably less than for source current." Adcom runs the LTI0I0 at very high quiescent currents (the quiescent currents of the LTI0I0 can be set by an external resistor) to reduce this effect. Another problem disclosed in the data book is that the excess phase of the LTI0I0 can be large enough, when driving capacitive loads, to cause stability problems. To improve stability the buffer must be removed from the feedback loop at high frequencies by an RC network in the feedback loop. This results in increased distortion at high frequencies. A simple complementary output stage formed with two or four low-cost discrete transistors and a couple of diodes as part of a discrete op-amp (see Table 1) would appear to be a simpler, more cost-effective approach than the use of the LTI0I0 and an IC op-amp. In the Adcom preamp the LTI0I0 chips are mounted on large heat sinks to improve reliability of the devices.

Measured distortion of the line stage at low frequencies was excellent. With the outputs unloaded, the 1 kHz THD plus noise reached a minimum of -93 dB at 9.5 V rms. At higher frequencies the results were less good, with 20 kHz distortion reaching a minimum of only -80 dB at a small 2 V rms. Adding a 600 ohm load to the output did not significantly affect these results. The less than optimal 20 kHz performance would be expected, based on the analysis above. An additional source of dynamic distortion in integrated circuits that use JFETs in the differential pair (such as the LT1056, LT1022, and LT1122) has been identified by Walt Jung [Jung 1992]. In a standard IC process, the JFETs are isolated from other parts of the chip by an isolation well. The isolation well forms a nonlinear parasitic capacitor to the substrate of the chip. Jung reports that the value of this capacitor varies (since it is nonlinear) with the common-mode voltage, and this gives rise to a dynamic distortion mechanism when the input source impedance is high (as is the case when the volume control is at normal levels). In our tests the volume control was set to full scale, so only a 500-ohm protection resistor was in series with the input; thus Mr. Jung's new findings should not account for the 20 kHz THD performance. Channel separation was flat at -105 dB from 20Hz to 100Hz. It then rose at a 6 dB per octave rate. At 20 kHz the separation was an excellent -65dB.

The phono stage of the Adcom is formed with a single gain stage. The RIAA network is implemented in the feedback loop of the op-amp. Adcom claims the phono stage is direct-coupled, but the RIAA network uses a large 2200 µF electrolytic capacitor (see sidebar) for C_2. This capacitor is bypassed with a 4.7 µF film capacitor. Adcom claims that the RIAA network is a special low-impedance design which reduces noise. In reality this network has an impedance characteristic very similar to what we find in the Sumo and Acurus, and it is an order of magnitude *higher* in impedance than the network used in the B&K. The Adcom has buffered tape-monitor outputs. This is an important feature which is not found in the Acurus, B&K, and Bryston. A powered-down tape recorder can present a very low, nonlinear input impedance. A tape-monitor buffer decouples this load from the source driving the preamp. It is very important that the phono stage of the Adcom not be loaded because it has a relatively high output impedance (Table 1) caused by the passive RC network used to cancel a parasitic zero. This problem was discussed above.

Measured performance of the phono stage was very good. RIAA equalization error was less than ±0.1 dB. The imbalance between the channels was remarkably small, since it never exceeded 0.025 dB. The worse-channel input-referred noise level was 0.44 µV. At tape monitor out, the 1 kHz THD plus noise reached a minimum of -86 dB at 10 V rms. Again, the 20 kHz results were less good. The 20 kHz distortion curve started to deviate from the 1 kHz curve at only 1 V rms. From 1 V to 2 V rms the 20 kHz distortion was a rather high -70 dB. A minimum of -76 dB was reached at 6 V rms. Phono overload is no problem, since the output of the phono stage clipped at 10 V rms at all frequencies .

In summary, the Adcom GFP-565 is not a bad preamp but it is not as good a value as some other preamps in this survey. To make this more graphic—if B&K and Acurus used the same profit margin that I estimate Adcom uses, B&K would have to charge $1050 for the PRO-10MC and Acurus would have to charge $850 for the L10.

B&K *Sonata Series* PRO-10MC

B&K Components, Ltd., 1971 Abbott Road, Buffalo, NY 14218-3241. PRO-10MC (Sonata Series) preamplifier, $698.00. Tested sample on loan from manufacturer.

If the President of the United States wants to see an example of a consumer electronics company competing head on with the Japanese and winning, he should consider a trip to Buffalo, New York. At its price point the B&K PRO-10MC exceeds any competitive foreign-made product in construction quality and value.

Parts quality is excellent. The parts are autoinserted into the PC board and wave-soldered—B&K is no "mom and pop" operation. For the audiophile mystic it may be significant that B&K uses the same F-Dyne capacitors found in megabuck preamps. I do not particularly care what brand of capacitor B&K uses, but I do consider it very significant that this preamp has a double-sided PC board with plated-through holes, something you do not often find on the megabuck units. Mechanical construction is likewise excellent. Unlike all other preamps except the UltrAmp in this survey, the B&K has the RCA jacks directly mounted to the rear panel. The other units use PC-mounted RCA jacks. In the B&K, a large PC board runs the length of the rear panel. The hot sides of the RCA jacks are connected directly to this PC board. A pair of ribbon cables then route the input and output signals to the main PC board. The ribbon cables are connected to the main PC board through a pair of 28-pin IC plugs and sockets. This allows the main PC board to be easily removed from the unit if service is required. These connectors are not gold- or silver-plated and represent a small long-term reliability problem. Note that B&K marches to a different drummer, and they place the right-channel jacks above the left-channel jacks. Given the construction quality of this unit, I was surprised to find that the channel separation was substandard. The separation curve rose at a constant 6 dB per octave from -93 dB at 20 Hz to -37 dB at 20 kHz.

A total of seven inputs can be connected to this preamp. Balanced line drivers are optional. The balanced circuitry is op-amp-based and not up to the performance of the unbalanced line outputs. Save $200.00 and use the unbalanced outputs. The volume and balance controls are high-quality, sealed, laser-trimmed Nobel units. The volume control has mechanical detents that limit the ability to set the volume optimally. I prefer the undetented form of the Nobel pot used by Acurus and Sumo. The switches are silver-plated.

Ed Mutka, the design manager for B&K, has done a superb job in the design of this preamp's active stage. Identical gain stages are used for both the phono and line amps. The 13-transistor discrete op-amp has a topology (see Table 1) very similar to the circuits found on the Boulder, Coda, Spectral, and Threshold preamps. The B&K topology has the advantage of very high CMRR (common-mode rejection ratio) and PSRR (power-supply rejection ratio). This comes about from the use of a differential pair in the second gain stage with an active current mirror and a supply-voltage-insensitive current source to bias the input stage's differential pair. Proponents of fully symmetrical designs would argue that the mirror pole unbalances the signal symmetry at the input to the second gain stage, but I think this is insignificant for an audio-frequency amplifier. A pair of bootstrapped cascode devices linearize the input stage and reduce the capacitance seen at the input of the gain stage. The only significant difference between the line amplifiers in the megabuck preamps and the B&K circuit is the power rating of the output transistors and the value of the quiescent current that the transistors run at. I do not find this to have any practical disadvantage since the 6 mA quiescent current of the B&K ensures that the output stage will remain in class A when driving a 2.5-kilohm load at 10 volts rms. I have never seen a power amplifier with an input impedance lower than 10 kilohms. It is possible that the stage could be driven into the class AB region if it were required to drive large capacitive loads (greater than 5 nF) presented by long runs of loony audiophile interconnect cable. The solution here is to replace the loony cable, not the preamp.

Measured linearity performance of the line stage was excellent. With the outputs unloaded, the 1 kHz THD plus noise in the less good channel reached a minimum of -93 dB at 15 V rms. The 20 kHz distortion reached a minimum of -91 dB at 10 V rms. Adding a 600 ohm load to the output did not significantly affect the low-frequency results, but the 20 kHz distortion was slightly affected above 5 V rms.

No output coupling capacitors are used in this design. A DC servo nulls the output offset. The DC servo correction range is limited, and coarse adjustment of offset is achieved with a high-quality, sealed trim pot. In addition, great care has also been taken in the design of the gain cell to ensure that its intrinsic offset is very low. In essence, the DC servo only corrects for long-term offset drift in the amplifier due to temperature changes or component aging. Limiting the required correction range of the DC servo allows the loading of the DC servo on the feedback network to be significantly reduced. This also limits the offset at the output of the preamp to 300 mV under conditions of servo-loop failure. Normally a servo-loop failure can result in the full power-supply voltage appearing at the output. This in turn could destroy your power amplifier and speakers. The use of FET inputs eliminates input bias currents and hence the principal need for input coupling capacitors (C_1 in the sidebar). Despite the excellence of this approach, some residual DC must be "floating around" this unit since clicks were heard when the tape monitor, mono, and bypass function switches were activated or deactivated. Input coupling caps would eliminate the clicks by removing the DC signals at the switches' inputs.

The B&K's large toroidal power transformer is housed in a separate box to reduce hum, just as in the megabuck units. Power supplies are regulated by Darlington-connected pass transistors which have no global feedback. Heat sinks are used on the pass transistors. The bases of the pass transistors are connected to a zener diode which is biased by a current source. The current source improves the PSRR of the regulation in comparison with the resistive biasing arrangement used by Acurus. Both channels share the same regulators. The megabuck preamps typically use separate regulators for the left and right channels. The decision whether to use global feedback around the regulator stages is still a subject of debate among preamp designers. Using global feedback reduces the output impedance of the regulator at DC but degrades the regulator's stability. Krell and Coda use the open-loop design, so B&K is in good company. The unit does not mute the output on power interruptions. Turn-off transients are small, but the turn-on transient is more significant. It is not large enough to destroy most power amplifiers or speakers but it is large enough to strongly justify adding a muting circuit to this preamp.

Ergonomics of the unit are good, but I have a number of cavils. It is hard to get a good grip on the slippery-smooth, round control knobs. I would also prefer wider control knobs. My fingers kept bumping the tape monitor switch when I operated the function selector. The tape-monitor switch and function selector should be spaced further apart. The push-button switches tended to stick as they were deactivated; a better-grade switch should be used. The headphone jack was not well mounted, but I definitely prefer this to no headphone jack at all (as is the case for the Acurus, Bryston, PS Audio, Sumo).

The phono stage is implemented in a single gain stage (see Table 1). B&K uses an RIAA network that is a very low-impedance design, to reduce thermal noise. This requires the amplifier to be stable into a very difficult load, equivalent to a 50-ohm resistor in series with a 25-nF capacitor at high frequencies. If significant ultrasonic energy is present, the phono stage may driven into class B operation. B&K had made provisions to reduce this load by placing a resistor in series with the RIAA network, as done by Adcom, Citation, Sumo, and Threshold (see above and Table 1). However, the place where this resistor belongs is jumpered out on the PC board of the B&K. The approach was rejected on the grounds that "it did not sound as good." The electrical performance of the phono stage was significantly degraded by this decision. At the tape monitor outputs, the 20 Hz THD plus noise reaches a minimum of -80 dB at 5 V rms. The 20 kHz distortion reaches a minimum of -65 dB at 1 V rms. The distortion then increases rapidly as the output begins to clip. At 2 V rms the distortion is -43 dB and at 3 V rms it is -26 dB. This result confirms the above analysis that the circuit is unable to drive the RIAA network at high frequencies. With a cartridge that has a high output level this may be audible. Perhaps it was this soft clipping that was interpreted as "sounding better." The input-referred noise level of the phono stage measured a good 0.62 µV. RIAA equalization error was less than ±0.1 dB from 80 Hz to 20 kHz but it rose to +0.3 dB at 30 Hz. The imbalance between the channels never exceeded 0.07 dB. Given the good channel balance we can conclude that the equalization error in the bass range is not due to component variation and that it could be corrected by changing the value of one of components in the equalization network slightly.

A simple, two-transistor, open-loop head amp is included for moving-coil cartridges. I prefer the more complex closed-loop design found on the Sumo. The pre-preamp is switched in and out by a standard push-button switch which can be accessed only from inside the unit. I have some concern about the long-term reliability of this method of switching low-level phono signals. The gold-plated shorting link method used by Acurus would have been a better choice here.

This is clearly the "best buy" among the preamps tested. But it is not perfect. The tape-monitor outputs are unbuffered, an output muting circuit is missing, and the phono stage performance could be better. Unsealed silver-plated switches are a potential reliability problem, as are the IC pin connectors used to link the main PC board to the input PC board. The manufacturer offers only a three-year warranty on this product. Come on B&K, I bet you can design a product that would last 20 years for under $1000!

Bryston .5B

Bryston Ltd., 57 Westmore Drive, Rexdale, Ont., Canada M9V 3Y6. Model .5B preamplifier, $795.00. Tested sample on loan from manufacturer.

As Lee Iacocca would say, this is the best-built, best-backed preamp of the group. Bryston warrants this preamp for 20 years with unlimited mileage. Equally important is the fact that a substantial part of Bryston's business comes from the professional audio market. You cannot be a success in that market unless your products are bulletproof, with very low failure rates.

The process of opening this preamp tells you it's built to last. The top plate is held to the main chassis by screws from four sides, including the front of the unit. This provides greater rigidity. The sheet metal is very thick and very well machined. Inside the metal enclosure is a very well-laid-out PC board. The parts are of very high quality, and the leads are carefully formed. The board is single-sided but not a jumper can be seen. This is the result of a very careful layout and the use of a few 0-ohm resistors. The board does not have plated-through holes and it is hand-soldered. With this type of construction Bryston must inspect the solder work with extraordinary care if they want to keep returns small over the 20-year life of the warranty. All switches and potentiometers are soldered directly to the board. The PC-board-mounted power switch caused some slight board flexing when it was operated. Because the PC board is single-sided, the selector switch must be placed near the input jacks and it is driven from the front panel by a long metal shaft. This selector switch and other switches in this unit are thickly plated with gold. This is one way Bryston ensures a 20-year life for this unit. Gold-plated switches are very expensive. To see just how expensive they are, consider that the $555.00 price difference between the .5B preamp reviewed here and the Bryston 11B is primarily due to the change from a four-input selector switch to a six-input version in the 11 B and the addition of a record selector switch. The Bryston switch units are not sealed. Sealed switches would further enhance reliability but they are even more costly. The volume pot is a sealed, laser-trimmed Alps unit. The diameter of the resistor element is 3.5 cm. This is a full centimeter larger than the Nobel unit used by Acurus, B&K, and Sumo, affording a larger surface area for the wipers to contact the resistor element. This Alps pot has 12 wipers to further improve reliability. The balance control, on the other hand, is not of very good quality. It is a smaller, unsealed 2.5-cm Alps unit. According to Bryston, this pot will soon be replaced with a sealed Nobel unit. Acurus, B&K, and Sumo presently use this higher-quality pot.

The power-supply transformer is a large, high-current design. A 4700 µF capacitor is used for filtering each unregulated supply rail. Heat-sink-mounted, integrated-circuit regulators (7X24 series) are used to generate the 24-volt supply rails. The regulated rails are filtered with only a 50 µF capacitor. Using large capacitors can sometimes degrade the stability performance of some integrated regulators. Bryston uses a pair of small subregulators in each of the discrete amplifiers. Since the Bryston has six discrete amplifiers, the unit has a total of twelve subregulators. Only the supplies to the gain stages of the op-amp are subregulated. The output devices are connected to the unregulated rails. This method is claimed by Bryston to yield lower distortion. I see no reason to doubt this claim.

At first glance, the schematic of the discrete amplifier does not look very promising. The bipolar differential pair is not degenerated, and resistors rather than current sources are used to bias the differential pair. The the size of the biasing and load resistors of both gain stages does not appear optimal. Yet the distortion from these discrete amplifiers is virtually unmeasurable. What is going on here? Through years of refinement, Bryston discovered that many sources of second-order distortion could be canceled by seemingly minor circuit changes and very careful component matching. The result of all this work is a very simple, cost-effective circuit which performs as well as many more complex circuits but is more reliable, since it uses fewer parts. Keeping things simple is often the hallmark of a good analog design [Williams 1991]. Because the method by which these circuits work is not obvious, copying them is very difficult [Pease 1991].

Measured distortion performance of the line stage was excellent. In the "worse" channel, with the outputs unloaded, the 1 kHz THD plus noise reached a minimum of -91 dB at 9 V rms. The 20 kHz distortion reached a minimum of -88 dB at 6 V rms. Adding a 600 ohm load to the output did not significantly affect these results. Channel separation was substandard, rising at a constant 6 dB per octave from -90 dB at 20 Hz to -32 dB at 20 kHz. According to Bryston this unsatisfactory result is caused by coupling between PC board traces. The company says that a modified PC board with ground traces placed between the sensitive nodes is now in development.

In line with keeping things simple, capacitors rather than DC servos are used to remove DC offsets. Four capacitors are in the line-level signal path. Three of these are shown in Figure 1. The additional capacitor is placed between the input signal and the volume control. This prevents DC on input signals from causing noise when the volume and balance controls are adjusted. All capacitors except C_2 are high-quality film units. The type of polyester film capacitor used by Bryston is the same as found in many megabuck preamps. The output coupling capacitor (C_3) is rather small at 3.3 µF. The effect of the small capacitor can be seen when a 600 ohm load is added to the output. Under this condition the output level is down 11 dB at 20 kHz. C_2 is an unbypassed electrolytic. A power-on muting circuit with a time delay prevents turn-on and turn-off transients. Surprisingly, the tape-monitor outputs are not buffered.

Bryston was one of the originators of the concept of a two-stage phono equalizer. Bryston literature for the 1B preamp, dated 1984, clearly identifies all of the major problems associated with phono-stage design. Bryston has thus had seven years to refine this technology. The Bryston architecture is unique among the two-stage phono preamps because no passive equalization stages are used (see Table 1). The Bryston approach yields good noise performance and good dynamic-range scaling. Its main disadvantage is that the output of the phono equalizer is inverting. DC offsets are removed using the same capacitor configuration as the line stage.

The excellence of the design can be seen in the measured distortion performance of the phono stage, which bettered all other preamps in this survey. At the tape monitor outputs the 20 Hz THD plus noise reached a minimum of -89 dB at 9 V rms. The 20 kHz distortion reached a minimum of -85 dB at 7.5 V rms. Clipping at 20 kHz starts at 10 V rms, while the 1 kHz clipping level occurs slightly higher at 13 V rms. The reason for that is that the first stage of the phono section overloads before the second stage at 20 kHz. Since the clipping levels are very close, it can be concluded that the dynamic-range scaling of the two stages is close to optimum. RIAA equalization error was less than ±0.1 dB. The imbalance between the channels never exceeded 0.05 dB.

Bryston uses the same discrete amplifier in the phono stage as in the line stage. This amplifier uses symmetrical differential pairs as the first gain stage. That is not a very low-noise design approach (input-referred noise measured 0.73 µV), and a head amp or transformer is required for moving-coil cartridges, as neither is included in the .5B. Bryston does manufacture an optional transformer for use with the .5B.

This preamp is another low-silhouette Krell/Levinson quasi-look-alike. Ergonomics has never been a strong point of these pancake-style preamps. On the Bryston all of the controls are crammed together, and the shallow, round knobs are not easy to grasp. Because of the light weight and low profile of the preamp, it moves when the power switch is operated. Why this innovative company decided to copy others in this regard is beyond me. The original Bryston Model 1B, which had a larger enclosure, was a friendlier device to use. The .5B has only three line inputs, which is inadequate for many systems. I cannot understand why Bryston chose to duplicate the tape monitor input on the function selector, since another line-level input would otherwise have been available. The high gain of the line amplifier and the taper of the volume control required that the volume control be set at approximately the 9-o'clock position. This limited the ability to make small changes in volume level.

In summary, Bryston's under-$800 preamp can truly be expected to last 20 years, but it is a bare-bones design. Only by striping down the unit could Bryston achieve such high reliability at a three-figure price.

Citation 21 (follow-up)

Harman/Kardon Incorporated, a Harman International Company, 8380 Balboa Boulevard, Northridge, CA 91325. Citation 21 Control Preamplifier, $629.00. Tested sample on loan from manufacturer (last tested 1987-88).

This unit was reviewed favorably by the Editor in Issue No. 11, when it sold for $549.00. How does this American-designed, Japanese-built unit stack up at its new $629.00 price?

The preamp's construction is typical for an upmarket Japanese consumer producer. The PC board is single-sided with lots of jumpers. The board does not have plated-through holes. Parts quality is in many cases not as good as in the other units in this survey. For example, the resistors are carbon film types. More expensive metal-film resistors are used in the other preamps. The volume control is a high-quality sealed unit, but the balance control is a something out of a $200 receiver. The selector and power switches are mounted in the middle of the main PC board. They are driven from the front panel by long plastic shafts. Any small misalignment of the shafts through the front panel will cause them to bind. On the positive side, the unit has a nice-sized transformer, a pair of 4700 µF capacitors on the unregulated rails, and large heat sinks for the pass transistors.

The circuit design is clearly the work of an American high-end designer. It is is in many respects very sophisticated, but omission of important buffer circuits degrades the preamp's overall performance. Separate, discrete, closed-loop regulators independently drive the phono and line stage. This approach is common in megabuck preamps but it is not used in any other preamp in this survey. Additional subregulators are assigned to the moving-coil amplifiers and the headphone amplifiers. In all cases single regulators drive both channels.

The phono circuit is a two-stage design (see Table 1), using totally discrete op-amps. DC offsets are eliminated with electrolytic capacitors (see sidebar). The first stage of the phono equalizer is a transconductance (G_m) amplifier with an output impedance which could be over one megohm. The RIAA network loads this amplifier and sets the open-loop gain of the amplifier. The return feedback rate is kept constant using this approach, and the RIAA network also acts as the compensation network. An open-loop emitter follower buffers the output of the first stage. An open-loop buffer will exhibit higher distortion than a more complex buffer using global feedback.

As in the Adcom and Sumo, a passive RC network, placed in this case at the output of the buffer, cancels a parasitic zero to improve stability of the RIAA equalizer. As stated above, this results in a high output impedance. Unlike Adcom and Sumo, the Citation unit does not have tape monitor buffers, so performance of this stage will be degraded by cable capacitance and tape-recorder input-stage loading. You must power on all tape recorders connected to this preamp when using its phono input in order to reduce these loading problems. The moving-coil amplifier is an open-loop amplifier. It is switched in and out by a standard push-button switch controlled from the front panel. This is the same method used by B&K, and I have the same concerns about the reliability of this approach.

The line stage is a true high-end design (see Table 1) but it has a very high output impedance because it has no output stage: the output signal is taken off the second gain stage. Defeatable tone controls are built around the line amplifier. To prevent oscillations when the tone controls are operated, the tone control network is connected directly to the output of the second gain stage. This loads the output stage and reduces its open-loop gain when the load from the tone controls becomes reactive. This is the same technique used in the phono stage of the preamp; however, the line stage differs from the phono stage in that no buffer is used at the output of the discrete op-amp. I do not understand why this was done and I consider the omission of the buffer to be a very serious design error.

The Editor measured this preamp with our old lab equipment when it was originally reviewed. At 0.5 V rms no significant distortion from the line stage was observable on our old measurement setup. Outputs clipped at a somewhat low but more than adequate 8.5 V rms. Line-level channel separation measured -70 dB at 1 kHz but only -45 dB at 20 kHz. Phono equalization error appeared to be nil from 40 Hz on up, but the conservatively spec'd amplitude accuracy of the measurement equipment was only ±0.3 dB, and that could well have been the worst-case error, especially since a +0.3 dB error was in fact measured at 20 Hz.

The preamp lacks the "look and feel" of the North-American-produced units. At $549.00 you might just be able to overlook this but at the new price of $629.00 it becomes difficult. In any case, the high output impedances of both the unbuffered tape outputs and the line outputs prevent me from recommending this unit. Please note that the Citation power amps reviewed in Issue No. 11 do not have any significant design problems and continue to be very good choices at their price points.

PS Audio 6.0

PS Audio, Inc., P.O. Box 1119, Graver City, CA 93483. PS 6.0 line preamplifier, $599.00. Tested sample on loan from manufacturer.

The review that follows here is of a discontinued PS Audio product. They shipped the unit to me about 14 or 15 months before this article was finally ready to go to press. Since I wanted to cover the whole field in the one large article, the review was delayed. After I had finished evaluating the unit, but before laboratory testing, the unit broke down and failed to operate. We returned the unit to PS Audio to get a replacement. Randy Patton, the President of PS Audio, told me that I had evaluated an early production sample of the PS 6.0 and that later production addressed many of my complaints—and that, in any case, the preamp was about to be superseded by a new model called the PS 6.1. We were promised a 6.1 unit for testing but it never arrived.

When I called Mr. Patton to find out what happened, he informed me that he believed his company had been unfairly treated in the reviews of the PS Audio Digital Link II and the PS Audio Superlink (Issue No. 17, page 38) and that the company was not going to submit a 6.1. As it turns out, we made a factual error in stating that the Superlink was developed after Paul McGowan had left the company. The Superlink, like the 6.0, was designed during the final period of Paul McGowan's leadership at the company. *The Audio Critic* stands by its statement that the Superlink offers significantly less value in comparison with the Digital Link II. As can be seen from the review below, I found the 6.0 to offer less value than a previous PS Audio preamp, the 5.5. I suspect that the company was in serious financial trouble when the Superlink and the 6.0 were designed, and Paul McGowan attempted big cost reductions to raise profit margins. Mr. Patton took over the company shortly after this period. Mr. Patton states that the company is now in excellent financial condition. It is interesting to note that both products in question have been discontinued by PS Audio. Did they perceive that these products did not represent traditional PS Audio value? Since the PS 6.1 is derived from the 6.0, I feel my review is still relevant. Physically the 6.1 looks identical to the 6.0. It is unlikely to be a totally different preamp. Without a sample of the 6.1, however, I cannot comment on how many of the problems discussed below have been corrected.

Even before the above developments, this was going to be one of those negative reviews that manufacturers claim are the result of reviewer bias. To refute such a claim, let me point out at the start that I own a PS Audio 5.5 preamp and that after examining all the preamps in this survey I decided to keep my 5.5. The 6.0 does not have a phono stage or a record function selector. The removal of these features reduced the price of the PS 6.0 to $599. The discontinued 5.5—which was a full-function preamp)—sold for $1195 when last offered. When the $599 companion PS Phono Link phono preamp is added to the 6.0, the total cost is the same as (actually three dollars more than) that of the 5.5. In comparison with the 5.5, significant cost reductions can be seen in the 6.0. The RCA plugs in the back of the unit do not have gold-plated center terminals. The volume control is an inexpensive unsealed unit in contrast to the sealed Nobel pot on the 5.5. (All the other preamps reviewed here use much better-quality pots.) The balance control is so inexpensive that PS Audio supplies a front-panel switch to bypass it. Contrast this with the 5.5, which realized the balance control as a stepped attenuator. The thin sheet metal did not realign correctly after the 6.0 was disassembled. The PC board is single-sided, with numerous jumpers. The tape-monitor outputs, however, are buffered with LF353 op-amps in the PS 6.0. This is an improvement over the 5.5, which had no tape-monitor buffers.

An inexpensive button-type full-wave rectifier is used in the power supply. The unregulated side of the power supply rails passes through a proprietary power-supply filter bank. This circuit decouples the power lines connected to the regulator from the full-wave rectifier, in order to reduce the levels of rectifier switching transients and power-line transients which appear at the regulator input. Supply rails at 15 volts were the smallest of the group reviewed here. IC regulators (78M15 and 7915) are used, without heat sinks, in the power supply, and a single set of regulators drives both channels. The PS Audio 5.5 had a 20-amp full-wave rectifier, 30-volt regulated supply rails, an innovative six-transistor discrete regulator that used a MOSFET pass transistor, and a dual mono topology. The 5.5 did not have the novel power-supply filter bank.

The tape monitor, balance-control bypass, and active-stage bypass are activated by motionless switches. To achieve this stunt requires three relays, six SSI integrated circuits, 16 diodes, separate 5-volt and 12-volt power supplies, five transistors, and 55 passive components. The motionless switches are clearly more expensive to implement than regular switches and represent a potential reliability problem. I found the preamp would switch to balance-control bypass mode when I turned on a room air conditioner on the same AC line as the preamp. Your Editor had trouble activating the controls at all because of dry hands. (After this review was written, *Stereophile* reported that the similar controls on the PS Audio UltraLink activated unintentionally during electrical tests.) Clicks were heard when the unit's active-stage bypass was activated.

The PS Audio 5.5 had an excellent 10-transistor composite JFET/bipolar/power-MOSFET discrete line stage. In place of this sophisticated circuit, the PS 6.0 uses a composite circuit consisting of the Analog Devices AD847 op-amp and a three-transistor output stage. This AD847 is the same device as used in Digital Link. The output stage consists of a single-ended JFET source follower biased by a bipolar current source. All other preamps in this survey use push-pull output stages to reduce open-loop distortion. The AD847 is totally inappropriate for use as a line-level gain stage. Its high gain-bandwidth product and fast settling time are not required in a preamp line stage. The AD847 has high noise and low open-loop gain, both of which degrade measurable performance. The output is direct-coupled, and DC offset is nulled by means of a small PC-board-mounted trim pot. Drift in this pot is also a potential reliability problem. A front-panel switch will bypass the active electronics, connecting the volume control directly to the output. This results in a potential 12.5-kilohm output impedance (the volume pot is a 50-kilohm unit), which could cause a measurable and audible rolloff of high frequencies. A 2-µF film capacitor is used for DC blocking at the input of the line stage.

The 6.0 has no power switch. A relay shorts the outputs when power is removed from the unit. On restoration of power the unit defaults to the passive mode. No time-delay muting circuit is used in the 6.0, so care must be taken before switching to the active mode to allow the line stage to stabilize. It should be noted, however, that this is still an improvement over the PS Audio 5.5, which had no muting relay at all.

As I explained above, no measurements can be reported here because our sample of the 6.0 failed just before the Audio Precision tests were about to begin.

Sumo Athena II

Music Communication Systems, Inc., 9829 Independence Avenue, Chatsworth, CA 91311. Sumo Athena II modular preamplifier, \$828.00. (Without phono stage, \$679.00.) Tested sample on loan from manufacturer.

This is the preamp for the serious tape recorder user. The record function switch has separate switching for each tape recorder. This prevents the input and output of a tape recorder from being connected together. If that were allowed to happen, a potentially loudspeaker-destroying oscillation could occur. A very useful and innovative tape monitor switch is also included in this preamp. Normally a preamp that has a record selector does not have a tape monitor switch. You monitor the tape recorder's output by setting the input selector to the tape position. The problem with this is that it is very hard to compare the tape output and source quickly enough because the source may be several detents away from the tape position on the selector switch. The Sumo solves this problem with the tape monitor switch. When you press the tape monitor switch on the Sumo, the signal on the record selector bus is connected to the line stage. You can thus quickly compare the signal on the record selector bus (the source) with the signal coming from the tape recorder (the tape position selected on the input selector).

Tape output buffers are included on this unit and they are of a fully discrete, symmetrical design without global feedback. Most preamp manufacturers pay little attention to the design of the tape monitor buffer and use simple integrated-circuit buffers. Measured performance of the Athena II buffer was impeccable. At 2 V rms out, we measured a THD-plus-noise figure of -100 dB at all frequencies. This figure is better than it would be for a line stage because the buffer supplies no gain. At higher output levels, however, the distortion rises rapidly because of the lack of global feedback. This is of no consequence in a tape output buffer but precludes separate measurement of the phono stage at tape out (see below).

The sheet metal of the Sumo is very thin. Replacing the cover of the preamp proved difficult because the sheet metal did not align well. The unit has a double-sided PC board which does not have plated-through holes. The top foil side is used for a ground plane and thus requires a large number of jumpers. Provisions on the main board allow an optional phono board or digital decoder board, but not both, to be added. Our test unit had the phono board. The volume and balance controls are high-quality, sealed Nobel units. The selector and record function switches are unsealed linear units. The switches are silver-plated. A robust-looking rotational-to-linear converter drives the switches directly. A good-sized toroidal transformer powers the Athena II. LM317 and LM337 voltage regulator ICs are used; the regulated voltage is a high 35 volts. The left and right channels have separate regulators. A time-delay muting circuit is another quality feature of this preamp.

The topology of the line stage is given in Table 1. It is essentially similar to the circuitry used in the Bryston and the Acurus but it adds current sources to bias the differential pairs. Both C_2 and C_3 (see sidebar) are electrolytic capacitors. C_3 is a high-quality nonpolar unit bypassed with a small film capacitor. C_1 is not included. DC input current required by the operational amplifier is partially canceled by matching the base currents of the symmetrical differential pairs connected to the input. Measured THD-plus-noise performance of the line stage was superb. With the outputs unloaded, the 1 kHz THD plus noise reached a minimum of -96 dB at 10 V rms. The 20 kHz distortion reached a minimum of -93 dB at 8 V RMS. Adding a 600 ohm load to the output did not significantly affect these results. Channel separation was good but not outstanding. It rose at a constant 6 dB per octave from -110 dB at 20 Hz to -54 dB at 20 kHz.

The phono equalizer amplifier is identical to the line stage. As mentioned above, the stage is most innovative. It is the only single-stage topology, to my knowledge, which addresses all the major problems in phono equalizer design. Measurements of the actual implementation, however, proved to be a mixed bag. Input-referred noise level measured a relatively high 0.95 µV. Because of the tape output buffer design (see above), we measured THD plus noise through the line stage rather than at tape out, with the line stage gain set at -6dB. The numbers that follow are adjusted to indicate the voltage level at the output of the phono stage.

THD plus noise reached a minimum of -83 dB at 14 V rms output, at both 1 kHz and 20 kHz. Even at 14 V rms this figure is principally dominated by noise, not harmonic components. Of particular interest is the fact that the distortion performance did not degrade at 20 kHz. This is the expected result for this topology, which uses constant return-loop gain and an input stage with large amounts of local feedback. Unexpectedly, the 20 Hz results were less good. The 20 Hz curve started to deviate from the 1 kHz curve at only 3 V rms, where the THD plus noise measured -69 dB. A minimum of -72 dB was reached at 7 V rms. The most likely explanation is that the return-loop gain is being reduced at low frequencies. This can only happen if the "dummy network" which shapes the open-loop gain of the stage is no longer in control. The loss of control probably occurs because the intrinsic output impedance of the second stage of the amplifier (set by the output impedance of the transistors used in the second stage) is less than the impedance of the dummy network. One method to correct the problem would be to us a cascoded second gain stage. A cascoded stage has significantly higher output resistance. Overload immunity of the Sumo phono stage is excellent, with clipping at all frequencies starting at 20 V rms.

RIAA equalization error was ±0.3 dB. Two separate problems account for this substandard result. At frequencies below 100 Hz both channels start to roll off. This is exactly what one would expect to happen if the return-loop gain were declining below 100 Hz. If my proposal for the cause of this rolloff is correct, then the modification discussed above would also fix the equalization error. The second problem was a 0.3 dB rise in level, in the right channel only, from 1 kHz to 10 kHz. The left channel showed only a small rise, and the imbalance between the channels went as high as 0.2 dB. Clearly, one or more of the components in the equalization network is not being controlled to tight enough tolerances.

The moving-coil stage is not some simple add-on but a complete closed-loop amplifier. A topology using a paralleled common-source input stage assures very low noise —at least on paper. Separate subregulators supply the moving-coil stage. As in the B&K and Citation, an unsealed switch selects the moving-coil option. This is a potential long-term reliability problem because very small voltage levels are being switched.

Again, the measurements of the moving-coil stage did not match expectations. Input-referred noise measured 0.87 µV in the noisier channel, only slightly lower than the noise of the moving-magnet stage! The plot thickens when one discovers that the original Sumo Athena preamp from which the Athena II evolved produced an input-referred noise of just 0.13 µV under the same conditions of measurement. (This result is from a review of the Athena that appeared in the August 1989 issue of *Audio.)* Jason Stoddard, the chief designer at Sumo, states that the moving-coil stage is unchanged between the two products. There may have been something wrong with our sample, although it had already been sent back to Sumo once to have the MC stage rechecked. The input-referred noise actually dropped when the input was open-circuited—hardly the behavior of a perfectly working unit. A correction will appear in the next issue if our sample turns out to have been untypical.

The Athena II is another low-silhouette, pancake-style preamp. To make room, the volume and balance controls are concentric, just as in an old Lafayette receiver. The front panel is made of plastic. Small LEDs indicate the position of the input and record selector switches. Small clicks were heard when the selector switch was moved. It is possible that voltage changes in the signal leads that drive the LED indicator circuit are coupling into the audio signals.

Overall, the Sumo Athena II is recommended, especially as a line-level preamplifier. It combines extraordinary flexibility with a very linear line stage. The trade-off, to meet the $679 price point, is that the unit is not as well built as some of the other preamps in this survey.

UltrAmp Line Amplifier

Mobile Fidelity Sound Lab, 105 Morris Street, Sebastopol, CA 95472. UltrAmp Line Amplifier, $1695.00 (originally introduced at $1295.00). Tested sample on loan from manufacturer.

The UltrAmp Series is a new set of electronics distributed by Mobile Fidelity Sound Lab. UltrAmp components are distributed directly to the consumer, bypassing the dealer. The UltrAmp Line Amplifier was introduced at $1295, but as we go to press its price is up to $1695 (including a set of bonus CDs from Mobile Fidelity). Since it is sold directly, MFSL claims it should compete with preamps sold through dealers at prices well up in the two thousands and then some. From its external appearance, it does not appear to be in the $2000 to $3000 price class. Sheet metal work is relatively thin. The flat faceplate with four equidistant knobs does not look ultraluxurious. The blue-colored silkscreening is not ultrareadable.

Some of the construction quality inside the unit is quite good. High-quality RCA jacks are directly mounted to the rear panel and not on the PC board. The circuit board is a large double-sided PC board with plated-through holes. All controls are mounted directly on the board. The selector and tape monitor switches are very expensive sealed units manufactured by Electroswitch. The contacts of the switches are silver-plated. Pots are sealed units manufactured by Clarostat. The diameter of these units is only 1.5 cm. As explained earlier, this reduces the available surface area for the wipers to contact the resistor element. On the positive side, the element is plastic and the external contacts to the element are on the opposite side from the wiper. These design features enhance the unit's reliability. When the element contacts are on the same side of the element as the wiper, it is possible for the wiper to be damaged if it engages the external contacts. (This can happen in the fully clockwise or counterclockwise positions of the control.)

The balance pot is not a dedicated design for this function but just a dual linear pot. As a result, the gain goes up in one channel and down in the other channel as the pot is rotated. It is argued by the designer that this is a *feature,* since acoustic power is approximately constant as the balance control is rotated. The pot adds a significant series impedance at the input of the line amplifier even when it is in the central position. A balance control is usually a special part dedicated to this function, in which case only half of the circumference of the pot is resistive. The other half is metallic, shorting the wiper to the output of the control. A mirror image of this control is used for the opposite channel. When the control is in the central position it is out of the circuit. Clarostat does not produce such a dedicated balance pot.

The UltrAmp preamplifier unit is completely dual mono. Two small, inexpensive-looking, unshielded transformers drive separate integrated diode bridges. Ground for each of the two channels is kept completely separated. 2000 μF capacitors are connected across the diode bridge. The bridges supply a 7818 (or 7918) regulator, which then drives a 7815 (or 7915) regulator. This double regulator improves the PSRR, but the resultant supply rails at 15 V are the lowest of the group here. Most preamps in the $2000-plus price class would have discrete voltage regulators and higher-voltage rails. Using dual power transformers to increase channel separation will result in very little crosstalk improvement, since the current draw of a typical class A amplifier is independent of signal size. Crosstalk is dominated by board layout and by crosstalk in the volume and balance potentiometers. UltrAmp has not paid much attention to these simple items. For example, a ground trace is not run between the left and right input signal lines on the PC board. As a result the measured channel separation was rather poor. It rose at a constant 6 dB per octave from -91 dB at 20 Hz to -32 dB at 20 kHz. (Three different samples—see below why we measured three—were virtually identical in this respect.) The decision not to use a dedicated balance potentiometer probably contributed to this sub-standard result.

Although the dual transformers do not help the crosstalk, they do introduce grounding problems. With the Audio Precision generator outputs floating, output-referred noise and hum levels in our original sample were extraordinarily high. No other preamp in this survey showed any problems when the signal generator was floated. Grounding the generator improved the situation significantly in one channel but actually made it slightly worse in the other; even so, the improved channel was still the worst in the survey. A second sample said to correct the problem did not require the generator to be grounded but measured only slightly better overall—and then it was identified as an early, uncorrected version still, shipped by mistake. A third sample—this time, so help us, the corrected version, they said—was then put through the entire Audio Precision protocol once again, and the resulting measurements are the ones reported here. The line stage, although considerably improved over the original sample, remained relatively the worst in THD-plus-noise performance of all models in the survey. (This despite the fact that the line stage has a gain of just 14 dB.) With the outputs unloaded, the 1 kHz THD plus noise reached a minimum of -88 dB at 5.5 V rms. The 20 kHz distortion reached a minimum of -80 dB at 2.5 V rms. All frequencies clipped at 7 V rms. Adding a 600 ohm load to the output degraded the already mediocre results. The 20 Hz and 1 kHz signals reached minima of-80 dB and started clipping softly at 2.3 V rms. The 20 kHz signal reached a minimum of -77 dB at 1.4 V rms, where soft clipping began.

I cannot speculate on the cause of the distortion problems above because no schematic was supplied to me at the time of this writing. Without a complete schematic it is not possible for me to explain why a complex circuit with 14 discrete transistors and one IC (the IC is used for a DC servo) performs so suboptimally. Originally UltrAmp would supply schematics only if we signed a nondisclosure agreement. We said fine, but no schematics were forthcoming. As this is being written the company has indicated a willingness to report more information, but it has not arrived at press time. The company did supply a 4-page paper written by designer Michael Yee. The paper contained enough factual inaccuracies to fill a complete "Hip Boots" column. Discussions with Mr. Yee revealed that some of the inaccuracies are merely the result of his trying to oversimplify the topics discussed in the papers. In the 4-page document Mr. Yee claims that traditional distortion measurements are not relevant. This comes as no surprise, given the measurement results above. He claims to have developed new theories on human hearing and new tests for the design. I asked if we could have information on these tests but was told that the tests are too difficult to set up and that in some cases the company wished to keep the tests a trade secret. Mr. Yee disclosed to me that in order to optimize his circuits so that they performed well on his new tests he was forced to compromise THD performance and output drive capability. Discussions with Mr. Yee proved him to be an intelligent, well-trained engineer, but at this point, based on the limited information supplied, I remain very skeptical of

Analog Devices' New IC Op-Amp for Audiophiles

I thought I would never see it but it's here—an integrated operational amplifier targeted for high-end audio equipment. The **Analog Devices AD797** represents a major advance. The op-amp settles to a full 16-bit resolution in under a microsecond. No other op-amp spec sheet has ever reported a 16-bit settling time. Noise of the op-amp is equivalent to a 50-ohm resistor from 10 Hz to 1 MHz. The AD797 can achieve distortion-plus-noise levels of -120 dB (1 part in a million) at 20 kHz and at 7 V rms while driving 600 ohms at unity gain. With a gain of 100, distortion plus noise is only -100 dB at 20 kHz and 3 V rms. These remarkable figures are achieved by the use of a fully complementary IC process previously used only for high-speed op-amps. These high-speed op-amps have high noise levels, high offset voltages, and high bias currents, compromising their performance in audio applications. Fully complementary processes are very expensive, and the price of this op-amp is more than 10 times the price of the ubiquitous 5534. This price is much too high for mid-fi applications; thus the available sockets for this op-amp are dramatically reduced. It is highly unlikely that enough sockets exist for this op-amp in the audio industry (that is why this has not been done before), so it has also been designed to be used for other applications requiring high precision and medium speed.

Process technology alone does not produce state-of-the-art circuits; it also requires a creative design engineer. Scott Wurcer, the designer of the AD797, has produced a number of significant innovations [Wurcer 1992], which were developed after a detailed mathematical analysis of the distortion mechanisms in op-amps (no tweako shenanigans here!). First, he developed a new bootstrapped folded cascode stage which has a DC gain of 134 dB. By eliminating multiple gain stages, the number of distortion-producing stages is reduced. This new stage requires extremely close matching of some of the transistors, so it cannot be realized in discrete form. Second, Scott developed a new distortion cancellation method to reduce distortion from the unity-gain output stage. Distortion from the output stage is higher in this op-amp than with a discrete approach because the output stage has a bias current only 500 µA. Scott's approach differs from the error correction circuits developed by Hawksford or nested feedback approaches developed by Cherry. The Wurcer method cancels the output stage error but it does not appear in the signal path. A frequency-domain [Goodenough 1992] analysis using ideal circuit blocks shows that the transfer characteristic of the output-stage block does not appear in the transfer function of the op-amp when the correction circuit is correctly implemented. Dynamic output impedance is also reduced using this technique. The complete AD797 is a very complex device with a total of 60 transistors.

Since this single op-amp was designed to fit the largest number of applications possible, some trade-offs had to be made. The design goals required that the op-amp have very low noise, reasonable power consumption, and unity-gain stability. To achieve all these goals, some parameters such as slew rate (18 V/µs) and V_{th} (0.036 V) had to be compromised. Wurcer's analysis shows that distortion from the input stage is not as significant at 20 kHz as one would expect given the very low V_{th} because the op-amp has a very large (80 MHz) gain-bandwidth product. A high gain-bandwidth product allows sufficient closed-loop gain at 20 kHz to linearize the distortion products of the input stage (the 5534 and LT1028 work similarly).

In line stage and CD player applications the noise from the feedback components dominates, and the op-amp can be allowed to be noisier. Op-amps with FET inputs or degenerated bipolar input stages can thus be used. These stages are more linear and thus have higher V_{th}. Op-amps using these input stage also have higher slew rates. Perhaps Analog Devices will add additional devices with different input stages in the future. They might also consider the use of a composite bipolar/JFET input stage developed by the PMI division of Analog Devices for the PMI OP-275. This stage attempts to combine the low noise of an non-degenerated bipolar input stage with the high V_{th} of JFETs.

Although the AD797 op-amp has extraordinary settling characteristics, the slow 18 V/µs slew rate may limit its use as an I/V converter. Slew rates of 2500 V/µs are now being achieved with another Analog Devices product, the AD811 transimpedance amplifier. The AD811 does not have the precision of the AD797, so its distortion is higher, and the AD811 can settle to only 12 bits. The AD797 will have significantly higher errors on current step transitions in comparison with the AD811. Subscriber Geoffrey Grieble points out that the finite rate of change of the current at the output of a DAC limits the improvement in these errors that can be achieved by using high-speed devices, so the actual improvement may be much less than the difference in the slew-rate numbers suggests. A novel I/V converter shown in the AD797 data sheet is claimed to have excellent performance. It will be interesting to see which chip will be found optimum by CD player designers.

It should be clear from the above that the AD797 is now the audio op-amp of choice. Any high-end designer who chooses an LT1028, LT1115, 5532, 5534, SSM-2139, LM883, OP-27, or OP-37 for a new design is now doing so to save parts costs. If you are paying a high price for a high-end component, you have the right to find only the highest-quality parts under the cover. It will be most interesting to see how designers of discrete op-amps respond to the AD797 challenge.

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his claims.

Please note that I take no pleasure from the mistakes of a young designer who is apparently trying to do an honest job. If UltrAnalog were a tiny company owned by Mr. Yee, I would have put the unit back in its box and sent it back without a review, in the hope that the next attempt would be better. (The Editor tells me that Kevin Voecks, of Snell Acoustics, was at least as tweaky as Michael Yee before he metamorphosed into one of this country's best and most levelheaded speaker designers.) But UltrAnalog is a division of Mobile Fidelity Sound Lab, not a micro company. MFSL clearly intends to sell a large number of these units, and we cannot simply sweep this item under the rug. I do not believe that the top management of MFSL has any idea that the product they are selling may be flawed.

Mr. Yee claimed that the results of his research would be clearly audible only if a system consisting of the three UltrAmp components (power amplifier, preamplifier, and D/A processor) and a minimum-phase loudspeaker system were used. Despite my extreme skepticism, the Editor spent half a day disassembling a reference system and setting up the all-UltrAmp system with the Thiel CS2.2 speakers to find if any differences existed. Upon turning on the system, the UltrAmp power amp exploded, almost taking out the Thiels in the process. Failure analysis as reported by Mr. Yee disclosed that the primary failure was that of an output transistor with an open bond wire connected to the base. Since this amp uses only one output transistor per supply rail, the signal connected to the base was forced to the positive supply rail, and internal components were damaged. It is impossible to conclude from one failure what the reliability of the product is, but I must note that an open bond wire can be a sign of excess output transistor die temperature. Another cause for concern in this power amp (also $1695, up from the introductory price of $1295) is the 50 V rating of the power-supply capacitors, which are handling 48 V. Not much margin there for the money. The amplifier failure prevented the Editor from reporting on the sound of the combined units, at least in this issue. (A new power amp was received at press time—the score is three samples of the preamp and two of the power amp thus far—so that a report can be expected in Issue No. 19.)

The preamp itself also came close to destroying *my* system with a large turnoff transient. I did not expect this, since a muting relay is included. Apparently the power supplies collapse so fast that the relay does not have a chance to close before the op-amps go unstable. Mr. Yee informed me that this problem will be corrected on the next revision of the circuit board, but he is unable to correct it on the present board. This is the type of slipshod engineering that most designers would try to hide from. Not Michael Yee, who puts his signature on the back of the unit and on its PC board.

After all of the above, the fact that the unit has a nice, discrete, 12-transistor tape-monitor buffer did not matter. More to the point is that, even if the design problems in this preamp were all corrected, it would still be only an average value *in a retail store* at $1295—never mind $1695! A comparison to a $2500 phono/line unit such as the Coda 01 is just silly. Where is the machined metal cabinet, the gold circuit board, the internally shielded toroidal transformer, the balanced line outputs, etc.? The UltrAmp Line Amplifier thus cannot be recommended at this point.

Recommendations

So which one should you buy? If you do not need a phono input and do not plan on keeping the preamp for 20 years, go for the $599 **Acurus L10.** Excellent ergonomics is one of the highlights of this unit. Electrical performance is also uniformly excellent. It might even last 20 years, but the manufacturer only guarantees it for two years. The only downside to this unit is that it does not have a power-up muting relay.

The **Sumo Athena II** in its line-level version ($679) can also be recommended. It is not built to quite the same quality standard as the Acurus but it does have a muting relay. Its tape monitor flexibility and buffered tape outputs make it the clear choice in this group for the serious tapist. The Athena II also has the most linear line stage. You can add a phono stage to the Athena II (total cost $828), but this stage gives only adequate performance in its present form. Another option for the Athena II is a D-to-A converter card. We have not tested this option, but combining a preamp with a D-to-A converter looks like a good cost-saving idea.

The $795 Bryston .5B is guaranteed by the manufacturer to last 20 years. (The **Bryston .4b** sells for $750. It is identical to the .5B but it has another line input instead of phono.) Both the line and phono performance of the .5B are outstanding, except for channel separation, which we found to be substandard. The main problem with the .5B is that it can be used only in simple systems.

The **B&K PRO-10MC** is a "best buy" at $698 if you need a phono stage. It can be used in more complex systems than the Bryston. It even has a headphone jack. On the downside, it has no muting relay, its channel separation is no better than the Bryston's, and the phono-stage performance is not on the same level with the Bryston. Although its build quality suggests a long life, the manufacturer guarantees it for only three years.

If you want everything, including a moving-coil cartridge transformer and a headphone jack, the **Bryston 12B** may be your best choice (review coming in the next issue). But it costs $1795 and it still does not have buffered tape-monitor outputs. Then again, very few people need everything, and most have no reason to spend more than about $800 for a phono preamp.

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(continued on page 63)

In Your Ear

"The midrange liquidity is very VTL-ish...

...the bass has the slam and speed of the Krell...

... the rhythm and pace are almost Kimber-like...

... but the soundstaging would be a lot better with the Tice power line conditioner."

Calderhead 1977 (rev. Aczel 1992)

The Delta-Sigma Approach to CD Playback: Five Major Examples

By Peter Aczel
Editor and Publisher

Does the "conservative" multibit DAC technology in CD players have a future? Maybe not, if the performance of these delta-sigma (Bitstream, MASH, etc.) units in standard tests is any indication.

A detailed discussion of low-bit, high-sampling-rate D/A conversion appeared in Issue No. 15 (pp. 19-22) as part of David Rich's long article on CD player technology. The terms delta-sigma and sigma-delta are currently being used (interchangeably, with little chance of standardization) to identify this branch of digital electronics, and commercial labels like Bitstream, MASH, and many others (the more alphabet-soupy the better) are being bandied about rather loosely. My purpose here is not to bring you up to date on the latest engineering thinking in this area and sort out the technical differences; Dr. Rich will do that much better than I could, quite possibly in the very next issue. All I am doing here is to report on the actual performance of delta-sigma equipment available for CD playback, especially since that performance is quite spectacular and may well influence an imminent buying decision.

The listening tests (sorry to disappoint you).

Theoretically, these five different pieces of equipment could have occasioned ten different ABX listening comparisons. I must confess that I didn't go nearly that far; I did, however, run a sufficient number of double-blind tests at matched levels to convince myself that all differences were well below the threshold of audibility. The possible exception is the Sony "Discman" D-303, which has an ever so slightly rolled-off high-frequency response (down about 0.25 dB at 12 kHz and 0.5 dB at 20 kHz), conceivably discernible on, say, cymbals or triangles to super ears—not mine. What the tweako subjective reviewers fail to grasp is that an audible difference must have a cause, some kind of mechanism whereby the difference can occur. When an audio signal path is as accurate on an I/O basis as it is in this equipment, such a mechanism doesn't exist. For example, to talk about "soundstaging" differences in equipment with the kind of channel separation, phase accuracy, and low-level resolution we have here is nothing short of fatuous. Of course, said tweako subjective reviewers are suddenly unable to zero in on such differences when the brand names and prices of the equipment are withheld from them.

Since I hate to be wrong, especially when it comes to cause and effect in listening tests, I should perhaps qualify the above with one small reservation. There *is* a distortion mechanism in delta-sigma modulators that standard tests may not reveal. In fact, David Rich's article pointed it out and devoted several paragraphs to it. So-called limit cycle oscillations can give rise to low-level tones in the baseband that are not musically related to the signal source. These are called idle channel tones and under certain conditions they *may* be audible. Proper architecture of the delta-sigma modulator and the use of the correct kind of dither will counteract this elusive form of distortion very effectively. I encountered no discernible evidence of idle channel tones in my own tests, but if some future tester should claim to have detected—or even heard!—such distortion in these or similar units, I wouldn't dismiss the report out of hand as tweako fantasizing. Our Technical Consultant, Steven Norsworthy of AT&T Bell Laboratories, who happens to be a leading delta-sigma specialist, promises to pursue the subject further in a future issue.

The measurement protocol.

Audio Precision published a very useful 40-page application note on CD player testing a couple of years ago, and I adhered fairly closely to the procedures presented in it. The test discs I used were the CBS CD-1 primarily, the Japan Audio Society "Audio Test CD-1" (YDDS-2) for its very convenient 31 full-scale 10-second spot frequencies, and the Philips "Audio Signals Disc 1" (SBC 429) for some of its specialized tracks. For testing the one outboard D/A converter included here (Theta), I used the S/PDIF output of the Audio Precision DSP generator.

I did not test the CD transport mechanisms by themselves because I know of no meaningful performance tests. Undoubtedly, some mechanisms are better built than others and will have a longer trouble-free life span—I'll comment on that within the individual reviews—but in terms of preserving the integrity of the digital code on the disc any transport that isn't total junk is as good as the next, tweako reviewers to the contrary notwithstanding. Jitter that affects the output of the CD player depends entirely on the latter's electronic circuitry, not the mechanical disc transport, so that tweako criteria based on phono turntable nostalgia are completely irrelevant. This is the digital age, and one of its glories is the elimination of the need for analog precision where already encoded data is being processed. As for the speed with which the laser pickup accesses the tracks (a Julian Hirsch specialty), I think it's about as important as the pop-up time of a toaster—I don't make toast and I don't play CDs when I'm in a hurry. (None of the four transports here is a slowpoke, in any case.) Lastly, David Rich points out that the famous calibrated defect-tracking tests on the Pierre Verany CD #2 are of limited relevance, since some of the best performers on those tests trip over banal little nicks and bubbles when playing production CDs, whereas the poor performers often sail through the same spots unruffled.

Marantz CD-11 Mk II

Marantz USA, a Division of Bang & Olufsen of America, Inc., 1150 Feehanville Drive, Mount Prospect, IL 60056. Model CD-11 Mk II compact disc player with remote control, $2500.00. Tested sample on loan from manufacturer.

A Philips by any other name is still a Philips, and this is their top of the line. Under their new North American marketing setup the expensive stuff is labeled Marantz (see, it has come full circle, Saul!), and the distributor is B&O.

The CD-11 Mk II is so far the most deluxe implementation in a complete CD player of the SAA7350 *cum* TDA1547 chip set discussed by David Rich in Issue No. 16. The TDA1547 is also known as DAC 7. This combination has been touted as a state-of-the-art contender, and not without reason, but as we shall see the Japanese haven't taken that lying down.

Externally the player looks identical to the Philips LHH500 (see Issue No. 16), continuing the cosmetic tradition—heavy machinery finished in gold—established with the LHH1000. Construction quality of the unit is as solid as of the latter two; when you heft it, its 37-pound weight tells the story. The chassis is die-cast, and so is the "CDM-4 Professional" single-beam swing-arm mechanism, which is Philips's best industrial-strength CD transport/optics and looks bulletproof to me. The data path is as follows: laser pickup to SAA7310 decoder to NPC SM5803 digital filter (20-bit output word length, 8-times oversampling) to SAA7350 Bitstream PDM device (3rd-order noise shaping, additional 24-times oversampling, on-chip DACs not used) to data inverter/separator to *two* TDA1547 single-bit DACs (each a differentia]-mode stereo DAC). The two DACs operate in dual differential mode. After D/A conversion, the analog output signal is available through direct-coupled outputs (phono jacks) as well as balanced outputs (XLR) via transformers.

The performance achieved with this highly refined design is truly outstanding, better than what I have measured in any multibit CD player, but not the best of the group reviewed here. Full-scale frequency response, de-emphasis error, and channel separation were so close to ideal as to require no discussion. Low-level linearity was also sheer perfection, and that includes the absence of harmonic blips above the 997 Hz tone on the -90 dB dithered track. Delta-sigma can do that. Full-scale THD + N versus frequency was very impressive, -93 to -94 dB across the entire audio spectrum, but that's still not quite 16-bit performance, even if we allow the analog output stage a tiny contribution to the total. Furthermore, such an excellent result was obtainable only through the unbalanced outputs; the transformer-coupled balanced outputs measured -81 to -82 dB, with some serious 60 Hz contribution evident. Thus the balanced outputs cannot be recommended. On the 1 kHz, -60 dB track, I measured a dynamic range of 96.8 dB, which is excellent but not the winner. With a full-scale 17 Hz tone exercising the DACs, pure quantization noise as measured through a 400 Hz highpass filter was -93.6 dB, which jibes with the above.

One very basic design problem in a delta-sigma system is out-of-band noise, and the Marantz shows some weakness in that respect. At 90 kHz the noise rises to -65 dB and interferes with certain inband performance tests below -60 dB, necessitating modified techniques. I have no evidence that this affects the playback of music in any way. Another slight peculiarity of the unit is that it inverts the recorded signal. (Absolute phase fanatics please take note.) As for RF energy at the output, it is at worst only 6 to 8 mV peak to peak, and I could barely identify it on a 100 MHz scope as being roughly in the 100 MHz band. Not very dangerous.

I was perfectly happy with the control functions and ergonomics of the Marantz, although generally I'm not in favor of having only the absolutely basic buttons on the front panel and everything else on the remote control unit. The latter, however, is very complete and very high-tech; it even has a jog dial and a shuttle ring so you can make like Mr. Spock and navigate the disc at warp speed.

All in all, a very classy CD player, although at its high price I can't forgive its small shortcomings too easily.

Pioneer Elite PD-75

Pioneer Electronics (USA) Inc., 2265 East 220th Street, P.O. Box 1720, Long Beach, CA 90801-1720. Elite PD-75 compact disc player with remote control, $1200.00. Tested sample on loan from manufacturer.

I want to state right up front that this unit—which some time ago replaced the "best buy" multibit PD-73 and has many good things going for it at a still reasonable price—seems to have a strange design flaw I find very hard to accept. As soon as I discovered the flaw I asked for a second sample, which then turned out to have the same problem to exactly the same degree. From 800 Hz on up, the PD-75 shows quite excellent THD + N versus frequency characteristics at full scale, averaging -95 dB in the better channel and -92 dB in the less good channel. Below 800 Hz, however, the distortion rises sharply, reaching -78 dB at 250 Hz in the less good channel and -83 dB at the same frequency in the better channel. Below 120 Hz the distortion flattens out again but never reaches the lows measured at the higher frequencies. Through the balanced outputs the situation remains the same. I have no idea what causes the aberration, and Pioneer has provided no enlightenment on the subject so far.

The most unusual feature of this unit is the upside-down "Stable Platter" drive mechanism, which requires the CD to be inserted with the label side *down;* the working parts are all located above the disc. The claims for this somewhat startling departure from conventional design have to do mainly with disc wobble, vibration frequencies and amplitudes, etc.; the question is, however, whether the 0's and 1's in the data stream are aware of these improvements (see above). To my mind the prima facie advantage of such an "Australian" transport (sorry about that) is that dirt from the disc can't fall on the laser. My first sample of the PD-75 actually had a little trouble with this fancy mechanism; the drawer creaked and complained on opening and closing, although the clamped disc then spun without any problem; my second sample was flawless in this respect. Other things being equal (but they never are!), the Stable Platter by itself would neither clinch nor break a sale of the PD-75 to me if I were a prospective buyer.

Once again, full-scale frequency response, de-emphasis error, and channel separation were too perfect to require comment. Low-level linearity was also textbook perfect, without the slightest deviation from the recorded signal; the 997 kHz tone on the -90 dB dithered track was squeaky-clean. High-level linearity, however, as shown in the 997 Hz THD + N versus level test from 0 dB to -90 dB, was much worse at -20 dB than at any other level (by 2 dB in the left channel, 4 dB in the right), and that I find a bit strange. On the 1 kHz, -60 dB track, dynamic range measured 94 dB in the less good channel, and the quantization noise measurement (see the Marantz review above) was -91 dB in that same channel. Delta-sigma can be better than that.

On the other hand, the wideband noise when playing digital silence looked very good, reaching a peak of -110 dB at 45 kHz. It is possible, however, that the circuit automatically mutes in the absence of a digital signal, in which case the out-of-band noise measurement is not valid for the playback of recorded tracks. RF energy at the output was minimal; the frequency was approximately 100 MHz and the amplitude 3 m V peak to peak in either channel.

I have no complaints whatsoever about the control functions and ergonomics of the PD-75 other than the fact that Pioneer, too, is now following the minimal-front-panel-maximal-remote-control trend. I guess I'm fighting a losing battle on that one. Overall, my opinion is that the Pioneer Elite PD-75 is a near miss in an "ultimate" delta-sigma CD player and that in its next incarnation it will most probably be superb.

Sony CDP-X779ES

Sony Corporation of America, Sony Drive, Park Ridge, NJ 07656. CDP-X779ES compact disc player with remote control, $1900.00. Tested sample on loan from manufacturer.

This may be the first review anywhere of Sony's new "flagship" CD player, as I was lucky enough to obtain a very early sample (so early that the instruction manual was still in Japanese). I might as well spare you the suspense and spill the beans right here and now—digitally, this is the most nearly perfect CD player known to me. It is capable of playing back a 16-bit CD with true 16-bit accuracy—or at least so close to 16 bits as to leave no room for quibbling—and no other player I know will do that.

Yes, there's a new technology involved. Three Sony engineers by the name of Toshihiko Masuda, Masaaki Ueki, and Hiromu Masaoka (why should only American and English designers of the tweako fringe get their names into the small audio journals?) came up with "a better mouse-trap," namely the CXD2562 pulse D/A converter, and now the famous Philips SAA7350/DAC 7 combination looks to me only like the silver medal contender—if I may mix my Emersonian and Olympian metaphors. The technical information released by Sony on their new device has so far been rather sketchy, noncommittal, and a little unclear; as far as I can tell it has a MASH-type architecture incorporating 3rd-order noise shaping, a 90 MHz (way above anyone else's) system clock, and some kind of anti-jitter circuit on the DAC chip itself. The theoretical dynamic range achievable with this new technology is claimed to be 131 dB (21.5-bit equivalent); that of course is not what happens in a real-world CD player, at least not outside the walls of ad agencies. In the X779ES, two of these new DAC chips are used in a dual differential configuration. There's also a new noise-shaping digital filter, the CXD2560, which precedes the CXD2562 pulse converter in the new Sony topology. As for the analog line amplifier, it uses a DC servo instead of coupling capacitors and includes some kind of FET stage in class A operation (it says here—although I was unable to see a complete discrete FET amplifier on the circuit board, and no schematic was available).

Another feature of the X779ES is a digital servo for optical tracking control; in other words, the analog signal from the laser pickup is A/D converted, processed in the digital domain, and then D/A converted for pickup control. I have some residual doubts about the tremendous advantages that this means, but it sure is high-tech. The disc transport is the best of Sony's G-base assemblies, constructed of a highly damped marble-like compound; the heavy, beam-reinforced

FB chassis is internally copper-plated and appears to be very sturdy; the whole unit weighs over 36 pounds. The brushed-aluminum external finish and simulated rosewood side pieces are cosmetically pleasing. As for control functions and general ergonomics, I never met a Sony I didn't like—to operate, that is—and this one is no exception. Everything clicks and slides smoothly and conveniently. You can even eliminate the preamp and effect volume changes via the remote control, but the 2.1 V full-scale output may prove to be insufficient. I must say again that I prefer a keypad on the front panel, but those days are over, it seems.

The Audio Precision measurements turned out to be quite astonishing. The full-scale THD + N versus frequency did not rise above -96.5 dB at any frequency in one channel and -97 dB in the other channel; at most frequencies it was actually a fraction lower than that. Now, -98.08 dB is the theoretically best figure achievable with 16-bit encoding, but when measuring through the line output of the CD player one must take into consideration the distortion and noise of the analog stage, which must necessarily be a little more than zero. Thus the X779ES is certainly within a hairsbreadth of recovering all 16 bits from the test CD. On the 997 Hz THD + N versus level test from 0 dB to -90 dB, the results were even more impressive: the curve just about hugged the -98 dB line at all levels from -10 dB on down. Furthermore, the distortion and noise characteristics of the player remain the same, within a small fraction of a dB, through the balanced outputs. Needless to say, full-scale frequency response, de-emphasis error, and channel separation were close enough to perfection to be left without comment. Low-level linearity measurements hugged the 0 dB line on all the tests (oops, there was a +0.2 dB error on the -100 dB dithered track in one channel but not the other—shocking!), and the 997 Hz tone on the -90 dB dithered track showed no discernible harmonics whatsoever. At higher levels the linearity was letter-perfect; dynamic range as measured on the 1 kHz, -60 dB track was -97.5 dB in the "worse" channel; quantization noise (see the Marantz review above for details of the test) was -97.2 dB in that same channel.

Wideband noise when playing digital silence was the lowest of all the players, between -148 dB and -128 dB from 30 Hz to 10 kHz, then rising to a maximum of -110 dB from 70 kHz to 200 kHz. Again, it could be that in the absence of a digital signal the circuit automatically mutes in the Japanese manner, just to fool us reviewers (whatever happened to Bushido?), but then the player also does pretty well on the foolproof tests, doesn't it? Square wave response is normal; the polarity test shows a noninverting characteristic; RF energy at the output is approximately 30 mV peak to peak, without any distinct frequency I could see on a 100 MHz scope. Radiated RF energy must be quite negligible because an FM tuner sitting about two feet from my usual CD player location was suddenly free from gurgles, hash, and birdies when I switched to the X779ES. That's not a small matter to some users.

What else can I say? Maybe I should observe that the X779ES represents a kind of test case for *The Audio Critic.* Here is an audio component that comes closer to totally accurate reproduction on an I/O basis than any other in its category, at a price only hundreds, rather than thousands, of dollars above that of comparable units. Its sonic advantages are moot; therefore at $12,000 I'd say the hell with it. At $1900 its accuracy and overall build quality get my vote. After I had completed all the tests, the Sony was the one I left in my system.

Sony D-303 "Discman"

Sony Corporation of America, Sony Drive, Park Ridge, NJ 07656. D-303 "Discman" portable compact disc player with headphones, $359.95. Tested sample on loan from owner.

This is the second Sony Discman to be reviewed in our pages, not because we take these small portable CD players very seriously as audiophile items, but to illustrate the power of digital technology, which is a great leveler of performance differences between the high-end and mid-fi domains. (That's why those exquisite audio snobs left over from the '60s and '70s hate everything digital.) The first thing I did with the D-303 when I got my hands on it was to insert it into a $21,000 preamp/amp/speaker chain. Lo and behold, it sounded exactly like the seven times constlier full-size CD player it had just replaced. That, of course, is an anecdotal observation of very limited value—see the preamble to this series of reviews for my more conclusive comments on the double-blind comparisons—but to me the lesson was clear. We live in a new era of sound reproduction, and the old rules are no longer applicable.

I'm not saying that the D-303 is comparable to a full-size quality player in terms of intrinsic value or audiophile satisfaction. It's not as solidly built; during a country road test it skipped whenever the car went over a bump; it probably has relatively limited longevity; there's no keypad and no wireless remote control (although, when the headphones are used, a rather neat little pod on the cable duplicates the control functions)—I could go on, but the fact remains that it's a delta-sigma digital device and its measured electrical performance was in some respects better than that of high-priced, full-size multibit players of a few years ago. In other words, it has the technical wherewithal to sounds as good as it does.

Full-scale frequency response was ever so slightly rolled off at the extremes: -0.25 dB at 20 Hz and 12 kHz, -0.5 dB at 20 kHz. De-emphasis error was +0.25 dB at 10 kHz in the less good channel, better at lower and higher frequencies but not perfect. These small frequency aberrations could conceivably be audible. Channel separation was not very impressive for a CD player but still more than adequate: about 80 dB up to 1 kHz, rising to 62 dB at 16 kHz. (Many expensive preamps are worse.) THD + N versus frequency at full scale was quite good in one channel (-89.5 dB to -82 dB, up to 12 kHz) and not so good in the other channel (3 dB to 9 dB worse, depending on frequency); at 16 kHz things fell apart in both channels (-72.5 dB). Even so, that would have been big, grown-up CD player performance just two or three years ago. (Incidentally, the poorer performance in one channel had nothing to do with 60 Hz problems, even with the external plug-in power supply.) Wideband noise on the digital-silence track was in some ways better than with the Marantz CD-11 Mk II (really!), rising to a peak of -73 dB at 150 kHz; however, there were multiple smaller peaks on the way up there in one channel, and the same type of interference with inband performance tests at low levels was experienced as in the case of the Marantz. I was able to determine, nevertheless, that low-level linearity was nearly as perfect as with the other delta-sigma units; apparently that's money in the bank when the designer takes that route. One channel was definitely noisier even inband than the other; the dynamic range test on the 1 kHz, -60 dB track yielded 89.3 dB and 96.3 dB (quite a difference), and the quantization noise measured -81.2 dB and -91.7 dB (an even bigger difference). Obviously, there's a quality control problem somewhere. RF energy at the output measured 25 mV peak to peak and was of a very low frequency, about 3.3 MHz. Full-scale square waves were somewhat clipped by the digital filter, but that was to be expected; the polarity test showed that the D-303 inverts the signal. (I never checked whether the headphones correct that inversion.)

Overall, I'm rather impressed by this delta-sigma Sony Discman. Obviously, someone forgot to tell it that it's only a toy and not allowed to impersonate a real CD player.

Theta DS Pro Prime

Theta Digital Corporation, 5330 Deny Avenue, Suite R, Agoura Hills, CA 91301. DS Pro Prime outboard D/A converter, $1250.00. Tested sample on loan from manufacturer.

Here we have something a little different, the product of a small American high-end manufacturer rather than of a mainstream European or Japanese colossus. Theta, I have noticed, is one of the *very* rare outfits to have (1) generally excellent engineering, based on science, (2) high-quality parts and construction commensurate with the high prices, and yet (3) the respect and even adulation of the tweako subjective reviewers, who usually prefer something with a little more charlatanry. Amazing. Maybe it's because Mike Moffat, who designs the Theta components, knows how to talk tweako jive to the golden-eared cultists of the audio press while remaining a hard-nosed engineer when he works alone on his schematics and PC boards.

At $1250, the DS Pro Prime is obviously a bid for the low end of the high-end D/A converter market, which goes up into the five-figure stratosphere at its other extreme. The unit is based on the Philips SAA7350 Bitstream device, which certainly represents a good, cost-effective approach to high performance. The on-chip DACs are used; it would seem that separate DAC 7's (a la Marantz) either weren't yet available when the design was finalized or didn't fit into the budget. I suspect the latter, since the elaborate programmable digital filter, which requires an expensive DSP chip, couldn't have left much room for other deluxe options. Apparently Mike Moffat refuses to have anything to do with mundane digital filter chips, and that intriguing bit of engineering elitism raises a basic question in this instance. An early review of the DS Pro Prime hailed it as an important first, introducing the marriage of programmable digital filtering to Bitstream D/A conversion. That's just plain bull. This is delta-sigma. A large part of the extensive DSP needed before D/A conversion is performed on the SAA7350 chip itself. That's a given and not negotiable. What is the benefit, then, of having the initial steps of the **DSP** performed by a costly programmable processor? It's like paying for first class on a plane and then ending up in a coach seat anyway. Okay, that's a very loose analogy, and I'm not even trying to criticize the design as it certainly isn't doing any harm to the signal; I'm just questioning Mike's budgetary priorities. (Isn't that better than ascribing superior "soundstaging" to DSP-based digital filters, like that fatuous early review?)

Looking under the cover of a piece of Theta equipment is always a pleasure, and this is no exception. Nice, clean PC board, stuffed with quality parts. I had no schematic available, but the digital data path and analog signal path are fairly obvious. The S/PDIF chip is the Yamaha YM3623B, which is still the workhorse of the industry and is used here with additional performance-enhancing circuitry. Then comes the DSP filter, taking up lots of real estate on the board; then the SAA7350 with on-chip DACs activated; then an LT1028 analog op-amp in each channel (good, but see the sidebar on page 37—surely Mike will soon switch to the AD797); then an AD707 in each channel for DC servo (you didn't think Mike would use a coupling capacitor?); then an LM6321 output buffer in each channel (nice, high-speed, fully complementary chip). In other words, the silicon on the board is by and large a class act.

The Audio Precision measurements painted a picture that was very good but not amazing. The full-scale frequency response rolls off to -0.1 dB at 12 kHz and -0.4 dB at 20 kHz. I'm not sure whether that's an attempt at a teensy bit of top-end softening or just something that comes with the territory. THD + N versus frequency at full scale measured -92.5 dB across the board, at just about all frequencies. The worst-case exception was -91 dB at 8 kHz in one channel. That's excellent performance but still not quite up to the top-of-the-line Sony or Marantz standard. Linearity was perfect from the 0 dB level down to -70 dB, then off by +0.2 dB at the -80 dB level and by +0.7 dB at the -90 dB level. A 1 kHz, -90 dB fundamental showed a very clean spectrum, so I wouldn't attribute too much significance to those fractional amplitude errors. Wideband noise at the output with a digital zero input peaked in the 100 kHz to 200 kHz octave, at -80 dB referenced to 2 V rms. That's better than the Marantz result with the same Bitstream chip. A full-scale square wave displayed no visible clipping.

Radiated RF was once again a Theta quirk, causing some minor FM interference. Still, this is a very solid "basic black" D/A processor, which will be a good platform for further tests as we fine-tune our delta-sigma criteria. •

Interviewing the Best Interviewees in Audio Part I

By David Ranada
Contributing Editor at Large

After all the cult-magazine interviews with untutored audio amateurs who design tweaky tube amplifiers, what you need is the corrective effect of these dialogues with world-class professionals.

The following interviews *[meaning both Part I and Part II, originally not split up—Ed]* are the first of a series conducted with movers and shakers of the audio industry. These people were selected because they each have contributed an immense amount not only to high-end audio but also to "lower" categories of equipment on the sound-reproduction pyramid—right down to basic systems whose sound quality affects every listener, since they form an integral part of the signals being transmitted.

Most of these interviews were conducted by telephone. Nearly all followed the same general outline: the individual's history in the audio industry (starting as far back as recalled) followed by an assessment of the future paths audio might take. Although none of the interviewees was informed of the identities or the responses of other participants, there do seem to be some opinions of the future held in common by many of those questioned.

The most important of these beliefs—important because it points most directly to where much advanced audio research is going—is nearly universal dissatisfaction with good old two-channel stereo as it has been practiced for three decades. Further channels are now a compelling necessity for improved sonic realism. This engineering-driven demand contrasts sharply with the quad days of the '70s, when the extra channels were seen generally as two more opportunities to make money. Just how many more channels are necessary and technically feasible might surprise you.

As a corollary to this desire for more channels, many of the interviewees expressed wholehearted acceptance of digital ambience-enhancement devices as adjuncts to creating realistic-sounding reproduction. From the strength of some of the opinions, I get the feeling that if you haven't seriously investigated what such equipment can add to your enjoyment of recordings, you shouldn't consider yourself an audiophile.

Another important impression you will get from these interviews, perhaps the most important one, is the importance of properly conducted critical listening tests. Indeed, you will find out exactly how important critical listening is to people whose livelihoods—and the livelihoods of many others—depend on it. No casual weekend listening followed by writing a report for these guys!

Editor's Note:

There are basically two major categories of readers of any audio publication. There are those who want to learn more about audio and those who want to be told what to buy. Yes, of course, the two groups overlap sometimes and blend into each other but they represent two very different attitudes. The interviews that follow will be right up the alley of the first group, but I want to make sure that the second group doesn't skim over them either. After reading the informed opinions of these highly knowledgeable practitioners, you'll have a much better idea what equipment to buy and what to avoid than the wide-eyed readers of silly underground reviews of CD rings and power line conditioners. Trust me. Dig in and read

1. Interview with John Eargle, Recording Engineer

RANADA: You've had a very long career in audio, covering many aspects of the business. Have you ever thought of doing anything else?
EARGLE: I think audio is something that's basically in your blood. And you find out that [as] all the world changes around you, and as audio becomes a shrinking part of this world of technology, there's nothing else I could really do.
RANADA: Are you interested in video?
EARGLE: I don't enjoy video that much. I do enjoy going to a good movie. Most of my moviegoing is, quite frankly, at the Academy of Motion Picture Arts and Sciences, of which I am an associate member.
RANADA: So you get to vote on the Oscars?
EARGLE: Associates don't vote. But the Goldwyn Theater is probably the best theater in the world in both electroacoustical and optical quality. They maintain it beautifully because it is the showcase of the industry. And that's *the* place to see a good movie. Other than that, much of my time at home is spent listening.
RANADA: Can you describe your system?
EARGLE: Sure. JBL 250Ti speakers (which are a modification of the original 250, they came out about 8 years ago) and an 18-inch subwoofer to go with them. I'm powering them [the 250s] with a Sumo Andromeda, which is a very nice-sounding amplifier. The front end of the system is one of the current-model Philips CD players going through the passive volume control of the amplifier.
RANADA: What kind of music do you listen to?
EARGLE: Basically classical. I do a lot of critiquing of my own recordings, hearing in them little things I wish I had heard at the time [I made them]. And I do a lot of listening to other people's recordings. I'm not saying that I listen to other recordings per se. I listen to music, the [kind of] music I prefer. But I really can't take my thinking off how well or how poorly the recording has been made. I have a DAT machine now, and all the stuff that we record for Delos—after it gets edited—I bring home on a DAT copy for my final approval. I recently got a Lexicon 300, which is sort of a general-purpose digital effects generator. You can switch to a mode of operation which is a stereo manipulation device, like a preamp. You can change balance digitally. In other words, [you can] come out of the AES/EBU output of the DAT machine into the 300 and feed the analog outputs of the 300 directly into the amplifier. You can vary gain digitally; you have a variety of shelving equalization curves, and a few little other fixits that Dave Griesinger came up with, like crossfeed of low frequencies out of phase to give you a little bit more spread. It's a very versatile device. If I hear anything in any of my recordings that I want to change at the last moment, I have the privilege of doing it in the digital domain. Every now and then you may want to make a slight change of equalization.
RANADA: So there are some of your recordings that have been processed through this device?
EARGLE: Not yet, because the device is fairly new in my hands. There'll be a few things that will have gone through it, just as a matter of fine-tuning levels and balances from band to band.
RANADA: Do you find this transparent enough?
EARGLE: The thing is that you're in the digital domain, and the transparency you have really derives from the first [analog-to-digital] conversion process, and in the home, of course, from the last conversion process [from digital to analog].
RANADA: You obviously seem to be willing to work with digital technology even at its present state of development.
EARGLE: I'm very happy to say that most of the problems of digital have been dispatched—or essentially dispatched—in recent times. There are so many improvements in converters being made today that I really haven't got a problem with it. In a purely intellectual sense, I think we all wish that we had a higher sampling frequency, just for the sake of maybe that very, very tiny percentage of the population who can still hear a difference at 20 kHz. I

"...I feel that the...low-bit conversion techniques, noise-shaping methods, really do create a superior medium... one which I'm confident, despite what some people say, can be cloned ad infinitum."

don't really object to the 16-bit word length because we now have a very benign way of handling the bottom end of the scale. That's by dithering the signal and getting almost analog-like performance at the low end [of the dynamic range], where the signal can be heard to fade into a noise floor without any abrupt things going on, and no more of this nonsense that we heard earlier of the signal disappearing. You know, the old stuff about the reverberation disappearing when it got down to the least significant bit. I doubt that ever really happened anyhow because there's always been dither in the form of amplifier/preamp noise or even room [noise]: That's not ideal dither, but it can be effective enough to keep some of these things from happening. In any event, I feel that the quality of conversion today—the low-bit conversion techniques, noise-shaping methods—really do create a superior medium, one that I have absolutely no problems with and one which I'm confident—despite what some people say—can be cloned ad infinitum. As long as all the errors are detected and corrected, there's no reason cloning can't go on forever.

RANADA: I think that people who say that the reverberation disappears and whatnot probably don't go to many live concerts to hear what live sound does. Live sound actually has less apparent reverberation than you commonly hear on recordings.
EARGLE: Absolutely.
RANADA: The main difference is that in home playback you have only two speakers from where the reverb is pouring out. In a live concert you have surround sound.
EARGLE: Surround sound of a sort. You know, in most concert halls the amount of sound coming at you from the sides is high enough to be significant in the sense that you'd be aware if it were taken out. But you're not really aware of something coming at you from the side or from the back, normally. It's below a certain threshold of being noticeable as an adjunct to the performance. But it's very definitely there.
RANADA: When you engineer a recording do you try to include some of that reflected sound from the side?
EARGLE: Yes, yes. In almost every room in which I've made an orchestral recording—there's only been one exception; that was St. John's at Smith Square in London, which was quite reverberant, believe me, the way it was—I normally have a "house" pair of microphones, which are usually cardioids spaced about 10 to 15 feet [apart] and back about 25 feet. Each mike is pointing toward the side that it's on and slightly toward the rear. The idea is to pick up a predominantly house-related "signature" for reverberation that then gets panned hard-left and hard-right into the program. Now there's enough reverberation leaking into the front mikes to give you a good spread of reverberation from left to right. I'm not injecting a pool of reverberation at the left speaker and the right speaker. I'm really sort of enhancing what's coming in the main mikes up front. It's a very valuable ingredient in determining the effective distance from the orchestra that you want to operate. For example, in the recordings of the Grofé *Grand Canyon Suite* and the two Copland works [Delos DE 3104], we went for a little closer-in sound on the Copland and a little more spread-out sound on the Grofé. And yet it was basically the same mike setup. It was basically a matter of balance, purely and simply. That's what you can do with subtle, 3-dB balance changes.

RANADA: What do you think of the place of artificial reverb in classical recordings? It's very, very popular now with certain European labels.
EARGLE: If you have a good room you don't really need an artificial reverberator, normally. But let me tell you when it's absolutely essential. Look at the following situation where we have used artificial reverberation. In a recording of a soloist with an orchestra—the recording of János Starker doing the Kaddish by David Diamond [Delos DE 3103]—we had to have a mike on the soloist in order to have the right balance. There was no way he could be picked up by the main pair of microphones and be heard during tutti passages. The only way we could preserve the [sonic] "distance" we wanted on him was to mike him. That produced a sound like he was in front of the orchestra. So we added artificial reverberation to his channel only to match what might have been coming in from a player

who had been playing a little more forcefully. In other words, we were elevating his level, but [in order] to match the texture—the balance between direct and reverberant sound—we had to add reverb to his channel only. When I say "to his channel," [I mean to say that] it was fed from his channel, but we used a stereo reverberation device to get the spread that we needed. And that way you hear him sounding in relation to the orchestra the way he should—not as an appendage hanging out on a clothesline in front of the orchestra.

RANADA: A lot of the European producers are essentially eliminating the natural sound of the hall by coming in very close with a lot of microphones and reverberating the entire result.

EARGLE: I find that very, very hard to do convincingly. You can usually hear that. The only time I could see that being a necessity would be under the duress of having to record maybe three or four concerts, the way Bernstein was recording for Deutsche Grammophon during his last years—he insisted on doing [live] concerts. The only way you can get away from audience noises is to go in close, as you've described, and make a multichannel recording. You need to go multitrack at that point because you *will* have balance problems that you've got to correct later. You can't solve all those on the fly. With that many mikes going, it's really hard to do. All you can do at that point is to record everything close in and later create your own stereo image, your own stereo stage, by putting artificial reverb on it. I don't think I'd enjoy doing that. I guess I would do it if somebody paid me to do it, but that isn't the way I normally prefer to do business.

RANADA: You've been lucky enough to be able to call the shots?

EARGLE: That's right. And I've been lucky enough to work with companies who'd rather go two-track as a matter of philosophy and a matter of economics, because it saves a lot of money if you can do it direct-to-stereo and bump it over to the 1630 [digital mastering] format to do the editing.

RANADA: You obviously listen with a very analytical ear to your own recordings. How do you listen to other peoples recordings? Do you apply the same criteria as when listening to your recordings, or do you accept their different viewpoints?

EARGLE: I accept as artistic variation the fact that no two engineers are alike, and simply listen for what other people are doing. I can usually spot artificial reverberation, because I have a good ear for it (I've been around it for such a long time), and I'm very concerned about reverberant signatures in recordings anyhow. And I just like to analyze what other people are doing. I also like to analyze how other people manipulate dynamic range. There's some pretty artful work going on out there with recordings that sound like they have a fairly wide dynamic range but are really manually compressed.

RANADA: You don't do any of that?

EARGLE: I do very little and the little that I do is of the following sort: A loud movement or a fast movement of a work may be done [recorded] at one level, and the slow movement may be raised overall about 3 dB or something like that. That's about the extent of it.

RANADA: You're not riding gain during the music?

EARGLE: As a rule, no. But there may be an occasional situation where something might be missing. For example, if I have a spotlight mike on the harp I may raise the harp [level] ever so slightly just to be able to hear a little bit more. It's more just putting a little finesse on something than making it stick out.

RANADA: There's always a borderline...

EARGLE: It's a dangerous borderline; you want to steer well clear of that. Of course, if you do make a mistake, all isn't lost because you make that mistake once. And the producer looks at you and says, "Do you really want to do that?" and you say, "No" and back off a little bit the next time that passage is played. These are not one-shot deals. Normally a movement is played straight through. And then we go back and start at the beginning, and we record again up to a point where something happens—maybe an attack isn't right or a wrong note or whatever—then we back up and overlap several measures to a convenient starting point. We then overlap that and correct all these things. It's called "covering." Many

"...anybody's recording would sound better at home if there were a very high-quality, gently operated ambience system attached to it that would give you some early cues..."

times you end up with several versions of a difficult passage. Careful notes are taken by the producer; then the producer, who's normally the editor in the Delos mode of operation, [edits the tape]. It's at that point where I take the edited tape home and review it and then determine what, if anything, needs to be done. Overall level [is a concern]. [The peaks of] a recording should really top out somewhere in the top 4 or 5 dB of the range of the recording medium. Not that it's bad if it doesn't; it's just that people expect that to be the case, and there's no reason why you can't. We try to do that in the session itself. And if you operate in one place, like Seattle, as much and as long as I have, you've got all the settings written down. You simply walk right in and set everything up. And everything, from the downbeat on, is useful. You may make a fine midcourse correction during the first 45 seconds, or something like that, because of the nature of the music. But basically you are recording the minute the clock starts.

RANADA: Are you willing to experiment with new microphone techniques?

EARGLE: I'm willing to experiment with new *microphones,* let's put it that way. I never experiment with more than one variable at a time. In other words, if I want to experiment with a new converter, I don't want to try a new pair of microphones at the same time. Otherwise, if you hear a difference, what do you attribute it to? Microphones, of course, are far more variable than anybody's converters are. I don't do that much rampant experimenting because I've sort of zeroed in on some very nicely working arrangements with the two main pairs of mikes that I do use. The Sanken CU-41 microphones are in my estimation the world's best cardioid mikes. I can be very specific about this. In the ORTF array you use two cardioids roughly 8 inches apart and splayed roughly 110 degrees. (I normally splay them a little wider than that.) That means the middle of the orchestra, the winds primarily, will be coming into the microphones from well off axis. Most cardioid microphones roll off at high frequencies off axis. What this will give you in an ORTF pickup would be a rather dull sound in the middle of the orchestra. The CU-41s are the only microphones I have come across which very accurately measure exactly 6 dB at plus-or-minus 90 degrees [pickup angle] out to 12.5 kHz. All the other small-capsule microphones are down even by 8 kHz. So the Sankens, by virtue of their unique construction, work extremely well in the ORTF configuration. The other pair that I use is a flanking pair of omnis: the Sennheiser MKH-20s are certainly the quietest mikes I have ever come across. They have a noise floor of 10 dB(A), which is extremely low. If I go anywhere, those are the microphones I take with me. I'm willing to use anybody else's microphones for spot mikes, as long as they are high-quality microphones.

RANADA: What kind of mixer do you run them into?

EARGLE: Most of the Seattle work is done using Soundcraft gear. It's made in England—owned by the Harman group, for whom I do a lot of work. My own console is a sort of an experimental test bed for a lot of changes and so forth. It has nicely modified input circuitry on it and is a very, very quiet board. I will experiment with other stand-alone preamplifiers. But I normally use upward of 10 to 12 mikes in an orchestral recording session and I don't want to carry around 10 independent free-standing mike preamps without any way to combine them into stereo buses. You've got to have a console to do this.

RANADA: You obviously optimize your recordings for "standard" playback over a two-channel system. What are your feelings about home modification of the recorded sound, say by using ambience recovery or synthesis equipment?

EARGLE: I've always imagined that anybody's recording would sound better at home if there were a very high-quality, gently operated ambience system attached to it that would give you some early cues, from hard left and hard right, in the 25 to 30 millisecond range. (You don't really need much level when you do that.) And maybe a very gentle decay out the back. That could take the form of simply matrixing the recording, rolling off the high end, and maybe delaying it overall a little bit longer—not adding any reverb to it but simply delaying what is already present—

and running that out the back. Such things as that are fairly subtle. But if you listen to them and get used to them and then turn them off, you feel something has been taken from the playback environment.
RANADA: A lot of purists would say that that is essentially adding an inaccurate signal to the back. But you say it enhances the realism.
EARGLE: It enhances the realism because the notion of spaciousness is something that can be well defined. Heinrich Kuttruff, the German acoustician, said that spaciousness is derived from the following things: Sounds arriving predominantly from the side of the listener that are uncorrelated and that arrive within a certain time limit (not to exceed 50 milliseconds or so). The threshold of [these sounds'] audibility depends on the relative level. Primarily, they must be mutually incoherent—they should not be the same signal, which would of course give you kind of a center localization which would be terrible in a reverberant signal. I think that if you derive the [side] information purely from what is in the recording, you really aren't changing any of the basic balances. You're simply taking some of the cues in the recording and putting them back in the direction from whence they came originally.
RANADA: So your answer to the general question of realism vs. accuracy would put you in the realism camp?
EARGLE: I think so. I think accuracy is difficult to postulate with only a pair of channels, unless you're talking about a binaural situation where there is a direct mapping of one space into a psychoacoustic space in your head or, hopefully, outside your head.
RANADA: Advocates of the Blumlein recording technique [coincident crossed figure eights] would claim that it is inherently more accurate than others.
EARGLE: It probably is, in the sense of its being a canonic form that does give you even acoustical-power pickup in the azimuthal sense. But there are other problems with it. For instance, in order to get a good stage fill, the microphones have to be at an appropriate distance such that the angle subtended by the ensemble is 90 degrees. That sometimes puts you quite far away. I find that Blumlein techniques—crossed figure eights, that is—work best of all when you can arc the players so that they are all pretty much equidistant from the microphone. That's hard to do; you can't do that with an orchestra very easily. That's why it works extremely well in the right room with small ensembles. A solo piano can sound very, very good over a Blumlein pair, provided the room won't swamp it out by being too active itself. The Blumlein technique is the only technique that is truly canonical in this sense: it's mapping characteristics are really very, very good.
RANADA: But if you have to alter things in the recording setup to get good results, can it be all that canonical?
EARGLE: I think that we are looking at the exigencies of recording and what we have to do in this business. The problem is that one microphone hanging there cannot possibly pick up all the music that you want. It would pick up everything the same way. I'm firmly of the belief that the recording setup must be modified for the music at hand. I think that taking *Billy the Kid* by Copland and giving it the same treatment that you'd give a Schumann symphony is really a mistake.
RANADA: Some record companies try to make all their recordings, regardless of repertory, sound very much alike. That's against your philosophy?
EARGLE: It certainly isn't the way I like to do it.
RANADA: So you're not trying to create a "Delos sound"?
EARGLE: I'm not trying to create a Delos sound as such. I'm trying to serve the music the best way possible, which demands making these piece-by-piece changes. This is not casually done. There's always a meeting of the minds of the conductor, the producer and the engineer, and an agreement as to what kind of approach [should be used]. Now that we have so many of these things under out belt, we can say that we want the kind of sound we obtained on this record or that record, and we all agree. One thing I have done from the very beginning of this business is to keep fairly elaborate notes. I've got several engineering notebooks filled with mike arrays, console settings, microphone types, you name it. When we use a piano, I write down the serial number of the instrument. That's probably overdoing it. But I notate microphone locations, microphone heights, and special changes in the seating that might go with a given recording. We've gotten home a few times and found out that we need to do a passage better. So we do that passage the next time we're in Seattle (or whatever group it is). At the end of the session we'll take five minutes to go out and move the mikes if need be, reset the board for the way it was months ago, and make an insert. If you do that right, believe me, you can intercut measure by measure between what you've just done and what you did any amount of time earlier.

"I think the biggest variable at home right now is the loudspeaker. Electronics have reached a...very high level... You're nearing an asymptote and any...improvement comes with a lot of dollars."

RANADA: Where is the weakest link in the audio chain at the moment?
EARGLE: The stuff in the middle is what improves every year as we come to every AES [Audio Engineering Society] exhibition. The things that don't really improve ail the time are the recording techniques and the loudspeakers. The inputs and outputs of this vast system are where we are often in trouble. I think the biggest variable at home right now is the loudspeaker. Electronics have reached a very, very high level of performance. You're nearing an asymptote and any really significant improvement comes with a lot of dollars. The quality level of the electronics is just fine. We were talking earlier about CD players likewise reaching a very high plateau. That leaves the loudspeaker and the room environment. We're learning how to make really good loudspeakers. My own taste in loudspeakers for the home would be very, very smooth systems with devices that don't want to store much energy. In other words, tweeters that are extremely flat, peak-free systems, [and] well-damped rooms for the most part [with] a well-damped bottom end, and enough level capability just to give you a sense of realism. In the speaker of the future, I would never want to sacrifice accuracy for the ability to play loud. I find that as I get older I play things at lower levels, but still at fairly reasonable levels. I really think the outcome has to do with the choice of speakers, how the room is treated, and how much emphasis the listener wants to place on setting the speakers up properly. When you do all that, it's amazing—the clarity that can come through a system.
RANADA: Could DSP be used to solve speaker/room problems?
EARGLE: Frankly, I think I'd want to go in and treat the room physically. The only peaks and dips that you can't get rid of are those that are in the room. Those are predominantly below 500 Hz, where peaks and dips become apparent. There's something called the Schroeder frequency, above which the modal density in a room has reached a certain amount and is constant. In other words, above that point the room's transmission is fairly smooth. Below that point it begins to gather into individual room modes which have to be treated as such. In a very large room, like a concert hall, this occurs at very low frequencies. In the home you're talking about a frequency range of 200 Hz or so, below which these room effects become noticeable. Presumably some sort of DSP-based filtering system can be used to clean that up. But the cleanup that you do is only from the point you are measuring from, which presumably will be the listening position. I'd like to hear something like that work in my own home. I've heard demonstrations, but these have always been in hotel rooms at trade shows where there have always been other things to apologize for.
RANADA: Have you ever had the experience of having been overwhelmed by a system so that you've said that this is an amazing improvement?
EARGLE: Yes. Let me relate this. To the credit of the one person concerned, there's one loudspeaker I have heard in five different environments. And each time I've been there and played my own recordings, and I've sort of wiped my brow and said, "My god, that recording is better than I thought it was." All has been vindicated when it sounds that good over one system. The loudspeakers in question are the Duntech Black Knights. These are large vertical-array loudspeakers with a tweeter in the middle spreading out to the midranges. In other words, it's symmetrical in two planes, horizontal and vertical. These speakers are truly minimum-phase loudspeakers. That shows up in the published measurements. It

certainly shows up in the clarity of the systems. I guess that what I'm listening to here is not just the fact that they're minimum-phase or the fact that they are being run by very high-quality, high-current-capability amplifiers, but the totality of all this, and the care that has been taken in the setup of these speakers in every environment.

RANADA: Do you use these in monitoring?

EARGLE: No, I don't. They are extremely large, and you don't find them very often.

RANADA: Do you often listen to your recordings on other systems?

EARGLE: Oh yes. Right here in my right coat pocket I have a copy of *Engineer's Choice* [an Eargle-engineered Delos sampler, DE 3506]. It's just there for the occasional situation where somebody has a CD player and I hear something interesting enough that I'd like to hear a cut over their system. I feel that a proper recording ought to sound good over just about any system. Let me explain that. If you make a recording that only sounds good over a $50,000 loudspeaker system and sounds bad in an automobile, you're really missing the mark, because you're making recordings just for yourself or a handful of very rich friends. A recording really has to be aimed at a broad spectrum. Now I don't mean that you should compress it to the point where it has a dynamic range of 20 dB, and I don't think you've got to peak it up at the bottom end where many speakers may be shy of bass. But it does have to make a credible presentation over a wide array of systems.

RANADA: So you aren't doing anything special to get a versatile recording?

EARGLE: The basic ingredients of my recordings are the ORTF pickup with a little bit of stage widening that you get with the flanking mikes. Everything else is in the nature of a spot mike or a spot pair, and those are so down in level most of the time that they're really not detracting from the major pickup given by the four main mikes across the front. That approach, by the way, is sort of a modern translation of the Decca approach, where the ORTF pair has replaced the so-called left-center-right "Decca tree." The flanking omnis are really the same as the flanking omnis in the Decca array.

RANADA: The Decca tree is a classic recording technique that has never been adequately explained or written up. As I understand it, the mikes they use in the tree are not standard microphones.

EARGLE: The microphones they use are M-50s, which are omni microphones at low frequencies, but above about 1 or 2 kHz there begins to be a sharpening of the polar pattern towards the front and a slight rising in the on-axis response. It shelves out at about 6 dB hot at about 5 or 6 kHz and above. So it is directional at high frequencies and omnidirectional at low, which is why they have to be fairly widely spaced. Each one is about 1 meter apart on a triangle, with the left and right mikes on the base of the triangle, and at the apex of the triangle jutting forward a little bit, sort of over the conductor's head, is the center mike. And those are panned left-center-right.

RANADA: That has been the standard technique for Decca ever since the beginning of stereo.

EARGLE: It certainly has been their signature up to recent times. They're still doing it and making beautiful recordings. When they have a pattern for success it's foolish to just change.

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2. Interview with Roy Allison, Speaker Designer

RANADA: The first I had ever heard of you was in connection with AR [Acoustic Research]. But by that time you were already deep into the audio business. How did you get into that position?

ALLISON: The first professional position I had was as a staff writer at a trade magazine that served the mobile and point-to-point radio industries. It was called *Radio Communications.* I joined them in 1949 as a draftsman and a writer on the small staff. I was made an editor in 1951 and continued in that capacity until 1953. The same publishing company occasionally ran articles on audio, with an emphasis on audio quality. They always found that those were the profitable issues that made up for the others which were largely unprofitable. So they thought there may be room for a magazine devoted entirely to audio and music, and that was the genesis of *High Fidelity* magazine. I became a contributing editor in 1953 while continuing to be editor of the other trade magazines. I became associate editor in 1954 and audio editor in 1957. By that time the other magazines had gone, and we were concentrating entirely on *High Fidelity.*

"The first thing they did after we left was to develop the AR-8, a 'rock' speaker. I didn't think that was the way to go....They went downhill, and it didn't take long."

RANADA: Were you interested in audio before you started to work at the magazine?

ALLISON: I can't say that I was. While I was in the service, I got pretty intensive training in electronics. (I was an electronics technician's mate in the Navy.) And I built an FM radio with another guy, a sort of off-hours project. I started listening to music then. That sort of ignited a small spark, but not a big one. It wasn't really until I got into covering stories about audio and started working with the equipment that I really became interested. That lasted through 1959. I stayed with that publishing company—by that time the magazine had been bought by Billboard. I left in March of '59 and went to work for AR as an assistant to the president.

RANADA: This was not an engineering position?

ALLISON: Not at first. I wasn't hired to be in an engineering position. Because I knew all the magazine people, I was hired to be Eddie Villchur's assistant for PR. But there were problems in the manufacturing part of the company and I got involved in those, because part of my duties was answering customer correspondence. We had some peculiar problems. For example, we were just beginning to ship AR-2s. The systems were perfect when they went into the box, but by the time they arrived at the dealers the woofer cones were broken. There was a heck of a lot of head scratching about that. I finally discovered what the problem was: the box was too airtight. Any sudden drop or compression of one side would push the woofer cone in and break the apex because the coil bottomed against the magnet structure.

RANADA: This was in the early days of acoustic suspension speakers, and the AR-2 was the second speaker using the principle. So you were treading new ground with every model you introduced.

ALLISON: You're right, that's absolutely right. Anyway, I found the problem and worked out a cure for it. Immediately they thought that I must be some kind of engineering whiz. There were a few other problems I was able to take care of and I was gradually transferred from PR to engineering and worked in manufacturing as a production engineer. I got to be chief engineer in 1960, plant manager in 1964 (because my boss left), and in 1967 the company was sold to Teledyne, and I was given a 5-year contract and made vice president of engineering and manufacturing. While I was there, I set up AR's quality-control program and service policies. And I either designed myself or directed the development of the AR-3a, -4, -4x, -2x, -2ax, -5, -6, -7, and the LST. And I was also responsible for supervising the development of AR's electronic products. At the end of 1972, during the five years we were running the company after Eddie Villchur had left, sales had doubled and profits had doubled. But the size of the market had more than doubled. It was expanding extremely fast, especially at the low end. Teledyne wasn't happy with our losing market share and they fired the president (Abe Hoffmann) and brought in the guy who had been running BSR, who was a very nice guy, but who was totally, completely market-oriented. I said, well, it's time for me to leave, so I left at the end of 1972.

RANADA: How would you say your orientations differed?

ALLISON: I thought we should continue what we did best. The company was still profitable and still growing at an acceptable rate. And we had no experience at the low end of the market. We had no experience in making so-called "rock" speakers. The first thing they did after we left was to develop the AR-8, a "rock" speaker. I didn't think that was the way to go.

RANADA: Well, the market proved you correct.

ALLISON: Right. They went downhill, and it didn't take long. They went through an average of one new president every two years. But they've [alternately] recovered and slid back again ever since then. In the meantime, I spent a year working on the interactions of speakers with the room. By the time a year had gone by I thought perhaps I had something that would prove useful and viable in the market. We started Allison Acoustics in March of 1974, shipped our first product at the end of that year, and it was fairly successful for a while, most especially in the overseas market. We were leading the so-called high-end market in Italy and France for several years, also the UK for a couple of years.

RANADA: To what do you attribute this success, aside from the quality of your products?

ALLISON: For one thing we found good distributors, people who knew the market and who were skilled at exploiting it. The products were very successful there. The first five years I guess we hit our peak. After that we were never as successful as I thought the product merited. A series of recessions here and overseas left us floundering and we still aren't doing as well as we ought to do. Certainly we are more of a *succès d'estime* than a commercial success at this point.

RANADA: In terms of overseas markets, what kind of press have you been getting? In England the press has gotten very tweaky.

ALLISON: They have. Three years ago the original partners I had were bought out by a group of guys from AR, and there were a couple of guys who were with AR Limited in the UK. They also came with us and started a UK facility of Allison. They're making headway. They're doing reasonably well in the face of a severe recession, worse than it is here. (It used to be when we coughed, Europe caught pneumonia. But the world is getting smaller. When we go into recession it has almost immediate repercussions over there.)

RANADA: Have you made any headway in Japan at all?

ALLISON: None. Never have. We've had a couple of people who have tried, but we've never been able to crack Japan.

RANADA: What do you attribute that to? What makes the Japanese market so difficult?

ALLISON: I don't know. If I could answer that we'd be better off over there than we are. I know it's a huge market, and it offers an immense potential. But we haven't been able to do anything there.

RANADA: I want to go back to your earlier career and talk about the Electro-Voice lawsuit. Can you recap what that was about and what its repercussions were?

ALLISON: Eddie [Villchur] had a patent on the acoustic suspension system, which he wrote himself.

RANADA: It was his invention and his patent.

ALLISON: Yes it was, entirely his. He likes to learn to do technical things himself and he wrote the patent, and maybe that's one of the problems that he had later on.

RANADA: You mean he didn't have a patent lawyer write it for him?

ALLISON: No. He tells the story of going to see a patent lawyer and asking how much he charged, and the guy says, "$200 an hour," and Eddie says, "I'll take a half hour." He proceeded to go from there on his own.

RANADA: The patent was granted, I take it.

ALLISON: It was granted, and he actually had two licensees when I joined the company [AR]. One of them was Heath Company and the other was KLH. By the time I got there, Henry Kloss [the K of KLH] had already been gone two years.

RANADA: He [Kloss] was there [at AR] only a very short time, then.

ALLISON: Yes, he was. Eddie and he just were too disparate personalities. They were both company-president types and they couldn't both be president of the same company. They had a parting of the ways. But Henry took a license and actually paid royalties until Electro-Voice decided that they were going to make acoustic suspension systems and to hell with the patent. So they did.

RANADA: And didn't pay the licensing fees?

ALLISON: Right. Eventually AR sued Electro-Voice, and the suit had to be where Electro-Voice was, which was in Indiana. The court found that the patent was invalid because of prior disclosure on the part of Harry Olsen. He had described a system in which there was a very soft-suspension woofer in a small box. And Olsen described it and said that even in a small box this works pretty well. This was held to be prior disclosure, and AR lost the suit.

"...there [are] so many other things that go on in room measurements that it's hard to separate what the woofer is actually producing from what you measure at any particular point in the room."

RANADA: Do you agree with that conclusion?

ALLISON: I don't. I think that Harry Olsen, brilliant as he was, really had absolutely no idea what the implications were of such a system.

RANADA: He didn't mention reduction of distortion?

ALLISON: His emphasis was on [frequency] response, [that the] response was still pretty good in a small enclosure. Of course, Eddie completely described the reduction in distortion and the reason for it. I think that the patent laws are faulty in some respects. The concept and the commercial exploitation of the concept were obviously Eddie's and his alone. The fact that a device existed which could have been said to be prior art I don't think should invalidate that patent. But that was the interpretation of the law. And Eddie didn't fight it. He realized that a patent is useful in many respects but most especially for prestige. The real advantage of conceiving of something new and implementing it before somebody else is that you get a jump on the market and you learn more about it than anybody else knows at the time. It [a patent] is not of itself a guarantee of commercial success. And I can vouch for that.

RANADA: You have patents on your woofer, do you not?

ALLISON: Yes, I do. I also had a patent on the tweeter. And within a week after the patent was issued someone called to my attention that there had indeed been prior art which was not discovered in the [patent] search. So that tweeter patent now belongs to the people of the United States. That's what my patent attorney said to do: Make points for yourself. Really push it voluntarily.

RANADA: Do you know of people using aspects of your tweeter patent?

ALLISON: No, none that I know of.

RANADA: How about your woofer patent?

ALLISON: There are people who have designed around it but nobody so far as I know who has actually infringed. AR was one of the first, I think probably the first, to design around it with the AR-9.

RANADA: Could you recap what that patent was for and what led you to it? Loudspeaker/room interaction was rarely seriously addressed in those days.

ALLISON: Nobody thought of it, mainly because there were so many other things that go on in room measurements that it's hard to separate what the woofer is actually producing from what you measure at any particular point in the room. It's only because I made a large number of measurements and then averaged them that I was able to see that there indeed was a region of actually reduced woofer power output in a large number of AR-3s measured in actual living rooms. At the time I wondered what could have caused this. It was only after I left the company [AR] that I started seriously thinking about it and actually doing some measurements. And, lo and behold, reflections from room surfaces or hard boundaries really do reduce the output of a woofer in the frequency region where they [the woofers] are a quarter wavelength from the surfaces.

RANADA: This is not merely a listener-position phenomenon; the woofer output actually decreases.

ALLISON: Yes. It actually does decrease where one or more room surfaces are a quarter wavelength (approximately) from the center of the woofer. The effect is mild where only one boundary is concerned. But when more than one boundary—and in the worst case, three boundaries—are equally distant from the woofer, the woofer is effectively operating in a partial vacuum, which reduces its output by 10 dB or more.

RANADA: That's because the reflected sound waves alter the emissions from the speaker?

ALLISON: Yes, they're reducing the pressure on the surface of the woofer and reducing the radiation resistance because of that. On the other hand, when the woofer is a very small fraction of a wavelength from one or more boundaries, then the output is actually increased—doubled, quadrupled, or multiplied eightfold, depending on whether you have one, two, or three room surfaces.

RANADA: So this is the origin of the famed "Allison dip," which is a midbass

decrease in output.
ALLISON: Typically, where woofers are typically placed in a room, it's anywhere from 100 to 300 Hz.
RANADA: The response, if you have a good woofer, will rise below that point and above it, but right at that frequency there'll be a dip of several dB.
ALLISON: Yes.
RANADA: This happens to any woofer, regardless of technology?
ALLISON: Whatever.
RANADA: It's a basic property of woofers in corners.
ALLISON: Yes.
RANADA: And you solve this problem by doing what?
ALLISON: In the case of a three-way system, it's possible to solve it completely by getting the woofer close enough to the room surfaces and using a relatively low crossover frequency, so that the reflections are always inphase over the whole operating range of the woofer. So you get reinforcement and no cancellation. And then you put your midrange driver far enough away from these surfaces so that there's no quarter-wavelength effect within the operating band of the midrange—it's [actually] below the operating band of the midrange. With a three-way or four-way, that's possible to do, to solve it really elegantly. The most elegant solution of course would be a corner speaker with the woofer at the bottom of a triangular enclosure close to a room corner.
RANADA: Why do people ignore this problem when it seems to be a basic fact of physics?
ALLISON: I don't know; it seems pretty fundamental to me. But old habits are hard to replace. People are used to looking at systems which have the woofers mounted fairly high up on the front panel.
RANADA: If you measured one of your speakers in an anechoic chamber, it wouldn't measure as flat as some others at these frequencies.
ALLISON: No, it would not.
RANADA: Could that possibly be a factor?
ALLISON: The lure of the flat anechoic measurement. Mine would roll off at the low end [in an anechoic chamber].
RANADA: Put them properly in real rooms and the response would come up to flat?
ALLISON: Yep.

RANADA: How does one design around your patent? I thought a patent would cover all the angles.
ALLISON: You have to remember that devices that have been described before, even if they weren't done for the purpose that I described, still represent prior art. And if you take any woofer and put it on the side or on the top (in a two-way system), that was done before. The Swedes actually did that before I did. They didn't recommend putting it up against the wall, where it represents one very short distance. In a two-way system there is absolutely no way you can avoid negative reflected impedance into the woofer. No way; you can't get it close enough [at the high end of the woofer's operating range]. Something will happen within the operating range of the woofer because you have to work a woofer—in a two-way system—up to a kilohertz, at least. Now we said that the effect of multiple boundaries is not linear. The effect increases in a nonlinear way as you increase the number of the surfaces. If you have three equidistant surfaces, you get a huge notch. The way to avoid getting a huge notch obviously is to make the distance of the woofer to the three nearest surfaces as disparate as possible, as unequal as possible. And make as large a spread as possible. A good rule of thumb is to make the distance from the nearest surface times the distance to the farthest surface equal to the square of the intermediate distance. That's the geometric mean.
RANADA: I thought the golden mean was involved somewhere along the line.
ALLISON: The golden mean is about 1.6 to 1. When you talk about room proportions to equalize standing waves, the golden mean is a helpful rule. It's a good ratio. 1.25, 1.6, 2. But that has nothing to do with woofer placement. What you want is B squared to equal A times C. And then you have equally, geometrically spaced reductions in output but none of them very steep, especially if you can make C a large multiple of A. Now it's easiest to do that if you can make one distance as small as possible. You can do that by putting the woofer on the side or up top and putting it up close to the wall. So you have one very small distance and you have only two others to worry about. And that's a successful and easy way to make a relatively flat power [output] two-way system. But that is not patentable because it has been done, even though the Swedes don't recommend putting the speaker up against the wall. But they have made systems with woofers in the top surface. There's a serendipitous advantage to that, having little to do with power output. What it does is to make the woofer point away from the listening area so you can have a rising on-axis response and a flat off-axis response and still get reasonably flat power output from the woofer, so that your spectral balance is good in the crossover region and doesn't dip down the way it does when you have a flat on-axis response from a driver that becomes directional at the top of its range. So, as you walk around the system with the woofer mounted on the top, the balance doesn't change. I think that is helpful in getting a good, broad listening area in stereo. As I said, that's an area not covered by my patent. My patent [No. 3,983,333] has to do with basically three-way systems. The way AR got around my patent was very clever. They put two woofers, one on each side, close to the bottom of the cabinet. But not so close as to approach half of the minimum dimension of the panel, which also is part of the patent.

"California sound was motion-picture theater sound....you had to have relatively directional and high-level upper bass and midrange...The 'New England sound' did not have that tradition to cope with."

RANADA: There's another thing from ancient history—the late sixties—I'd like to go over, and that's the so-called New England vs. California loudspeaker-sound debate. Was there actually a New England sound, and how did AR react to the marketplace perception that there was?
ALLISON: California sound was motion-picture theater sound. The primary requirement for movies in those days, when the sound tracks were rather limited in frequency range, was intelligibility. Which meant that you had to have relatively directional and high-level upper bass and midrange sound. And I guess people used to bring those things [theater speakers] home, in their theater-size living rooms. And they were used to the sound and that was it. It sort of stuck. That kind of sound took root. The "New England sound" did not have that tradition to cope with. And the people who designed these speakers were free to try to make them sound as naturally balanced in a typical living room as they could remember live sound being in a concert hall. That meant, with recordings of those days, a tendency to have a tilted power-output slope (up at the low end and gradually down at the high end). That also was changing, as the quality of the program material has increased over the years. I don't think that there is that kind of differentiation today. I would say that now everybody pretty much shoots for a power output which is relatively flat, still a little bit tilted down but not nearly as much as it used to be. The reason is that the program material has so much less distortion than it used to. It's a much cleaner kind of sound and can stand a much more realistic balance in the mids and the highs.

RANADA: Are well-designed speakers sounding more alike these days?
ALLISON: It think it is much more so than it used to be. There are still pretty large differences to be heard, certainly larger than in any other audio component.
RANADA: How does a manufacturer pull himself above the crowd, now that speakers are beginning to sound more alike?
ALLISON: I guess you just try to do better than other people are doing. One of the things you can do is shoot for market niches. A lot of people have decided that that's the way to go and are going that way. Then there are the generalists such as Boston Acoustics who try to cover all of them. I try to make loudspeakers that look attractive as well as sound attractive and are good values. I think that we still have advantages that people recognize. I think that our tweeter, even though nobody else seems to want to copy it, is possibly the best way to go in terms of dispersion. I know that some people don't agree with that. Some people think that you should have limited dispersion in order to get firmer imaging or whatever. I think that the advantages of extremely wide dispersion far outweigh the disadvantages in terms of putting into the reverberant field all around you the same kind of balance of energy that you do get in a concert hall, except at the very low end, of course; you can't possibly duplicate the

low end of the concert hall.

RANADA: Recently there has been increased emphasis on what might be called nonstandard uses of speakers, in home theater and the like. I was wondering what you thought of this development?

ALLISON: I think it's very exciting. A good home theater system represents a different class of hearing experience than we've been able to achieve before. There have been related attempts to achieve similar kinds of things. Way back when I first started with AR, there was an organ company called Aeolian-Skinner. You remember their reverberation system? It was a special tape deck with—it must have been—a dozen playback heads [to provide multiple time delays]. It was used to make the typically nonreverberant, dead New England churches sound like large stone-walled cathedrals. It was pretty good for its day. It never caught on in a home application because I guess it was too elaborate or too complicated or too expensive. But it was dramatic, certainly, and successful in doing what it set out to do. There were reverberation systems from time to time that people came up with. They were interesting, but they never caught on. There was the quadraphonic phenomenon (or debacle), which might have been successful if we hadn't had the terrible conflict among the people who were trying to do it in three or four different ways. Later on there were ambience synthesis systems like the Sound Concepts and the Audio Pulse, which I thought were very nice. I'm still using the Sound Concepts in my home and have been for years and years. But if you get a good Dolby Surround system, or even a good music-surround system like the Yamaha [DSP] unit, it's addictive. At least I think it is. If we can solve the logistic problems of what to do with wires and multiple speakers and amplifiers, I think that it should catch on tremendously and should be very successful. We did introduce center-channel and satellite systems produced specifically for a home theater at the last [January 1992] CES. I have high hopes for it. I think it is a good way to go and is likely to produce much improved sound for a lot of people.

RANADA: Where do you see big progress being made in audio? What is there left to do in terms of speaker design or other aspects of the audio art?

ALLISON: I think that if the multiple-speaker system does bloom, as I think it will, we're going to see more and more miniaturization of the main speakers—the imaging speakers—and more and more separate woofer systems might be designed to fit under furniture as well as behind it.

RANADA: Do you think from your long experience that hi-fi has reached a point of stasis where the delivery media are changing but the actual sound quality is sort of stagnant?

ALLISON: I would agree with that.

RANADA: Where do you think the next big breakthrough will come, in terms of overall realism?

ALLISON: If it does come, it seems to me that it would be a logical development following from the present trend of more speakers, more small speakers located everywhere. And, of course, that almost requires different systems for recording. There's no reason these days why we have to be restricted to two channels in our recording media. There's no reason why we have to be restricted to derived four [channels] when we can have a real four. And if we have real four, then that multiplies tremendously the amount of accurate directionality you can get and accuracy in original sound-field reconstruction. It seems to me that if there's a breakthrough, it won't be sudden; it will be a logical progression of the direction in which we are now going.

RANADA: Were you involved with those live vs. recorded demonstrations at AR? Can you give me an account of what they were about and how they were useful beyond being good public relations?

ALLISON: They were recordings made outdoors of various musical groups. There was a [string] quartet, originally, and then a solo guitar. I think the final one I was involved with was an old nickelodeon. Recordings of these things were made anechoically. They were made on an experimental basis, trial and error. Trying to pick up the output of the instruments which would represent an average of the power delivered by the instrument. That was no easy task. We had to move the microphone around again

"If we can solve the logistic problems of what to do with wires and multiple speakers and amplifiers, I think [surround systems] should catch on tremendously and should be very successful."

and again and again. We tried different microphones and finally wound up with some marvelous new condenser microphones from Sony. They were the first really high-quality microphones that were relatively flat. The playback was always in a large room. It was never done in a living room because you couldn't fit enough people—get a large enough audience—in a small room. So they were done in large rooms where you might reasonably expect these instruments to be played live. And for that reason they were not subject to the same kind of low-frequency problems that you find in living-room reproduction. The speakers were far enough away from the room surfaces so that whatever aberrations there were produced by the room surfaces were very low in frequency, so that they were just out of the picture—so low that they were of no real concern. We found that we had to use a relatively simple tone-control equalization of the speakers' output in order to match the recording perfectly. There was some treble boost. The rationale for that was: everybody else had access to a treble tone control. And the fact is that the environment that they were played in led to much more high-frequency loss than you would have gotten in a living room. So you could say that it [the equalization] made up for that [the high frequency losses] and it really wasn't cheating in that sense.

RANADA: People could always point to the fact that you moved the microphone around in order to find the most advantageous sonic "loop" as cheating.

ALLISON: Of course. But unless you knew—for sure—exactly where to put it [the microphone] and then just put it there and recorded it, there was just no other way you could find the best location.

RANADA: That's because instruments do not radiate equally in all directinons.

ALLISON: Exactly, they certainly do not.

RANADA: So you have to find a location or direction when placing the microphone which, when reproduced over speakers, gives the same effect as the entire output of the instrument.

ALLISON: And if you believe that what you hear mostly in a reverberant environment is the power output of the speaker in that environment, then what you were searching for was the direct output [i.e., microphone location] that most closely matched the power output of the instrument. That was a problem, and it was a lot of work. But [the demonstrations] proved that it could be done with existing speakers even then. And the match was really exceptionally good. There certainly were times when you switched and you could hear a difference but you couldn't really tell what the switch was: live to recorded or vice versa. And that, I think, was about as close as you could possibly hope to get.

RANADA: Were the audience reactions positive?

ALLISON: Oh yes. Invariably enthusiastic. Eddie [Villchur] was a master showman, too. You have to give him that. He programmed this thing so that the switches were imperceptible most of the time and the only way you could tell that a switch had been made from live to recorded was when the musicians stopped playing. But after the switch they continued to play [i.e., pretended to] for a while, which helped the illusion tremendously. And there were inevitably gasps of amazement when the music went on and the bows stopped moving. The finale was a movement from the Mendelssohn Octet. Four parts were recorded and four parts were live—the [four quartet] musicians played with the recording throughout that whole piece. And of course you couldn't tell who was playing unless you watched the bowing extremely carefully and had the score in front of you. It was fun; it was really fun. And Eddie did it extremely well. I was really his gofer there; I made a few suggestions but it was really his show.

RANADA: How was your book written? [Allison, Roy F. 1962. *High-Fidelity Systems: A User's Guide.* Cambridge: Acoustic Research, Inc.]

ALLISON: Eddie said: "You gotta write a book. You have to do it. I've written one, and now you've got to tell people how to connect hi-fi systems." It was much more complicated at that time because we were just getting into the stereo era and people were obliged to use two [mono] preamplifiers, a master volume/switch control, and the wiring was kind of complicated at that point. And of course you had mono amplifiers and were obliged generally to buy a turntable and buy an arm and buy a cartridge all separately and put them together yourself, or have your friendly dealer do it for you, usually in a not very adequate manner. So a book was needed, and I wrote it on vacation time and weekends. And I think it did help some people. It was something that was really needed at the time and the first of the really quite objective how-to type books. After AR stopped producing it and selling it, it was picked up by Dover books. They sold it for quite a long time. They picked up Eddie's book, too. That might be still in print; I'm not sure.

RANADA: Looking back over your long career, I wonder if there's an accomplishment you are most proud of.

ALLISON: You count the number of years, and it seems like a tremendous length of time. On the other hand, to me it doesn't seem all that long. I don't feel as if as I should be thinking about retiring or anything like that. I'm still interested in this field. I still enjoy working in it; the challenges are still there.

RANADA: Is there an accomplishment you feel most proud of?

ALLISON: I think that the attention I paid to what the room does and the work that I did in that respect is probably the most important thing that I did.

RANADA: I'd agree with that. But it's amazing that you were almost pointing out the obvious.

ALLISON: That's the way that most "innovations" are. Once someone points them out you say, "Well, of course, everybody knows that." Except nobody said it before.

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3. Interview with Kevin Voecks, Speaker Designer

RANADA: So how did you get interested in audio?

VOECKS: I always wanted to be in it. When I was a kid, I did the usual thing of dissecting anything that resembled an audio component at home, then repairing other people's systems to make enough income in junior high school to supplement my parents' great expenses buying me hi-fi equipment. And then I started retailing it in high school—lots of equipment moving in and out of the garage while my mother was making sure the local officials didn't crack down. And that turned into Natural Sound [a retail store] that started out in Omaha and Lincoln [Nebraska]. Then I went to Worcester Tech in Massachusetts and started Natural Sound in Framingham.

RANADA: What did you study there?

VOECKS: Electrical engineering, which was specifically aimed at audio design.

RANADA: Did they have a program there in audio?

VOECKS: They claimed to. But in reality it's just that you could do major projects in whatever area you wanted to. It was like my high-school experience in that it was a somewhat experimental school and you could do projects to your heart's content rather than do the normal things you would do in school. That turned out to be incredibly valuable since that's all I do now: lots of projects, always a panic.

RANADA: What happened after school?

VOECKS: Our store was very successful, and it was during the right times of the '70s when audio was really booming, specifically high end, and there was very little competition so we had most of New England. But I was so frustrated that speakers were so far behind everything else. We had all the best of the best [in the store] and still the speakers were in general pretty sad. Peter Snell came into our store at one point and said, "I've got the world's best speaker" [the original Snell Type A]. You tend to ignore that—because lots of hobbyists at home think that because they have so much of their own time and energy involved in it that they really believe it. But he brought in these speakers that he wouldn't let us see. They were shrouded in grille cloth. We played them and listened half the night and we agreed that theywere the world's best speakers, period, end of discussion. He had worked on them while he was working for EPI. It was a unique combination of technology, in terms of seeing the bigger picture and looking at all kinds of first-order priorities, or things that ought to be first-order—like diffraction and like paying great attention to the effects of room boundaries at low frequencies and reflections from room boundaries over the rest of the range. I don't know and I don't think anyone will know exactly how he sorted out some of these issues without having access to the test equipment that existed at the time, let alone what's available now.

"One of the main changes was to move to higher-order crossovers from first-order crossovers when I was able to duplicate Stanley Lipshitz's findings on the relative inaudibility of phase..."

RANADA: It shows what a good ear could do.

VOECKS: But a well-guided good ear, one that can make the proper conclusions from listening tests. It's very, very easy to make the wrong conclusions even with the best listening. I had [started] the [speaker] company Simdex shortly after Peter started out in business and I did crude tests—which was all that was really possible at that point—to determine the audibility of having the drivers stepped back and whether it was significant that a speaker could reproduce a square wave. I concluded that it was and thought for many years that that was Peter's sole mistake, that he shouldn't have ignored the waveform and that he should have slightly rearranged his priorities. And it wasn't until years later [when I was] at Mirage that Stanley Lipshitz came up with a box that would allow you to adjust the phase response at will without altering the amplitude response. And once he made that, he came out with a paper saying yes, you couldn't hear them [phase anomalies] at mid and high frequencies even with gross amounts of phase shift. The only time you could hear it was in the midbass on specially selected clicks and things like that.

RANADA: What was the first speaker you designed?

VOECKS: The Simdex Sigma. And the first review for that was done by Sherri Lee of *Audiogram.* And if I am not mistaken the second review was by *The Audio Critic.* I remember taking the speakers to Peter's house with Andy Rappaport in tow. The speakers were well received. Then it was up to Canada to work for Mirage as a speaker designer.

RANADA: So you had only done one speaker when you went to work for Mirage?

VOECKS: Well, essentially. There were some others that we had at shows, like a system that required quadriamplification. *That* was practical.

RANADA: And at Mirage you did all sorts of things?

VOECKS: Actually they were the importer of Tangent, and Tangent suddenly needed to be replaced for a variety of reasons, so in a period of weeks a [speaker] line was designed and in production. Of course it changed as the years went on. One of the main changes was to move to higher-order crossovers from first-order crossovers when I was able to duplicate Stanley Lipshitz's findings on the relative inaudibility of phase and [when] all of the other advantages of higher-order crossovers were plainly evident. So that was an important shift.

RANADA: And after Mirage you took over at Snell?

VOECKS: Naturally I had continued to be close friends with Peter and greatly respected his work. About every year or so we at Mirage would look around for the best speaker we could buy to use as a reference standard. Again and again it was the latest version of the Snell Type A. He just never lost out on any point. Then Peter unexpectedly died of a heart attack. We were all very shocked. He was so dedicated to his company, and dealers had so much love for the product, even though they were getting very intermittent shipments and had to put up with no business policies at all at the company. They had so much respect for the integrity of the product that they wanted it to stay around. Fortunately, Dr. [William] Osgood bought the company and was easily able to convince me to come work for them. Carrying on Peter's work was very natural in that I had come around to everything he had believed at that point and didn't have to feel I was bending my beliefs in the least. I've always had absolute freedom—if I discovered that only green speakers sound good, that's what we'll make. It's an ideal situation.

RANADA: You now (late 1991) are deep in the middle of another speaker design?

VOECKS: At this show [the January 1992 Winter Consumer Electronics Show] we will be showing a slightly scaled-down version of the Type B [the B Minor].

RANADA: Are you going to the NRC labs [the National Research Council in Ottawa, Canada] for any measurements on this one?

VOECKS: I expect to do the final confirmations there. Fortunately the test equipment that is now available has advanced so much that it's rarely necessary at this point. I can make quasi-anechoic measurements with both MLSSA and the new LMS system. I can do power-response measurements as well.

RANADA: How important was the NRC in earlier years?

VOECKS: It was essential.

RANADA: What were they doing there that you couldn't do on your own and that nobody else was doing?

VOECKS: First of all, they had a very good anechoic chamber, one which was really dedicated to loudspeaker work in that even below the frequency at which it was anechoic, it was calibrated so that [one could measure] a forward-firing woofer's response down to at least 30 Hz. And the NRC had an automatic turntable system that would allow you to do power response and time-averaged window [measurements]. The second thing—and actually the more important one—was the work that Floyd Toole, Peter Schuck, and Sean Olive did in determining what the significant aspects were of what we heard. In other words, how to correlate real-world measurements with what you really hear. No one else had approached that very seriously, and they did it very rigorously over a period of 10 years. They have data that is not ambiguous at all, that shows you what actually must be optimized. It really helps you avoid falling into the trap of optimizing something that has a secondary effect at the expense of one that has a primary effect.

RANADA: Could you give me an example of this?

VOECKS: First-order crossovers with stepped-back drivers can have indeed better measured phase response, but the side effects [of such a design] impact these first-order things that we hear much more clearly, such as smooth frequency response from a lack of diffraction (it's hard to step back the drivers without having diffraction effects). Steep crossovers reduce the distortion and reduce the interaction between drivers so that the on-axis response can be flatter and in a wider window, and the power response is also optimized. Peter Snell had said since his earliest work that the power response, the reverberant field in the room, was as important as the direct sound, and I don't think anyone else said that for many years. *[See the Roy Allison interview in this issue.—DR]* Floyd Toole came to the conclusion that you needed to look at the far off-axis responses taken as a group because they would contribute to the reflections off the wall, floor, and the ceiling, and they would then contribute to the total radiated energy. So the sound you hear after the first 20 or 30 milliseconds is going to be composed of the off-axis response, not the direct sound from the speaker.

RANADA: You learned that from NRC. What have you learned from working with THX technology?

VOECKS: That's an interesting question. It hasn't contradicted anything. The original THX theater system was designed by using a pair of Snell Type A's in a relatively nearfield situation and a theater speaker system in the farfield. Tom Holman made a lot of really smart choices in doing that. So it's kind of come full circle; we're imitating the theater situation with other Snells. A requirement of the THX system is that it have a broad horizontal dispersion and that it be quite uniform. That is exactly what the NRC results said you should have. The THX requirements say that you must have intentionally limited vertical dispersion in the front speakers, since Tom's tests showed that increased the clarity of vocal material [dialogue]. When we went into this it was not clear whether following that rule in particular might not degrade the music, so it was a great relief that when we did double-blind listening tests we preferred the THX speakers over our Type C/II at the time. It was a surprise and it meant that there was no compromise to Snell's musical standards by optimizing for film material. His [Tomlinson Holman's] choice for the crossover systems was also a really good one too. I have never been a fan of

"When you think about it, 'taste' [in speakers] doesn't make sense because the very same ears hear a live performance and hear the speaker. Taste is factored out when it's accuracy you're after."

satellite/subwoofer systems because their integration was usually quite poor. Often the satellites would be designed with a bump in the low end so that they would be useful as stand-alone speakers. But that makes it impossible to cross them over properly to a subwoofer. Often the same manufacturer wouldn't make or specify the electronic crossover, so people were really free to mess that one up. Also, when they didn't know the existing rolloff of the satellite speaker there's no way you could graft a crossover on it and know what the combined result would be. So Tom specified that the satellite speaker will roll off at 80 Hz and that it would be a 2nd-ordcr rolloff so that it would be a sealed box. The electronics would have an additional 2nd-order electronic highpass and [the combination] ends up being a nice 4th-order rolloff. And since the woofer will of course play significantly higher than 80 Hz, there's a 4th-order electronic lowpass. It works in practice, since it is such a low crossover point you really don't have problems with directionality or midrange colorations from the woofers. It has the advantage over a full-range speaker that you can place the left and right speakers where they work best for imaging and put the woofers where there will be the best low-frequency response. We use two woofers because you can then get smoother response in the room, and again Floyd Toole's research published about a year ago shows how, by asymmetrically placing two woofers, you can dramatically improve the response at the listening position.

RANADA: Your use of two woofers is an example of the flexibility of the THX specifications. But I was wondering how strict they are. How much leeway do you have as a designer to make something that sounds different, or is the intention to make everything sound the same?

VOECKS: They are all supposed to sound at least as good as a certain performance level, and since you can't just say it has got to sound good, there are very detailed specifications that have to be met; for example, how flat the response must be within a 15-degree listening window. When I first read [the specs] my first impression was that the vast majority of high-end speakers would not meet that accuracy standard. That is still true. We went considerably beyond [the THX specs] and you can hear it. So the answer is that there is a range of sound quality within the specifications, although anything that meets them would be better than the pack.

RANADA: This goes back to some larger questions I wanted to touch on. For example, why is there such variability among competent loudspeaker designers as to what sounds good, even those who use the NRC as their reference lab? Their top-of-the-line speakers all sound different from each other.

VOECKS: Yes and no. Yes, they sound different from each other, but there are certain things that are in common. I think one of the most surprising things that the NRC's listening tests have shown is that all listeners—whether they are sophisticated listeners or experienced ones or somebody off the street—will prefer the same loudspeakers. The difference is that the skilled listener will zero in on [the good] one right away and will be consistent in saying so, and the naive listener will wander around a bit. But in the end they'll choose the same thing, even on a variety of music. This dispels a couple of myths; it dispels the idea that there are certain speakers for certain kinds of music, and the myth that it is just a matter of taste. When you think about it, "taste" doesn't make sense because the very same ears hear a live performance and hear the speaker. Taste is factored out when it's accuracy you're after. As you say, there are differences in the sound. For instance, the Mirage M-l is a bipole radiating from both sides, so certainly it will not sound the same as a Snell Type C/IV. But if you were to completely absorb the rear wave of the Mirage, I think you'd find that the two of them sound surprisingly similar.

RANADA: What role does taste play in your design? Or are you more instrumentation oriented? Even though you are known as a high-end designer, you seem to pay an inordinate amount of attention to instrumented results.

VOECKS: I believe in paying a lot of attention to that. You aren't doing your homework or being rigorous enough otherwise.

RANADA: And how does being rigorous

help you?
VOECKS: For one thing, it keeps you from falling into traps of making some wrong assumption or [of] spending money on products or areas that have no, or negligible, effect while there are other things that have been left undone that are important. For instance, even in a speaker at the price of a Type B we will look at cabinet vibrations and look at the threshold of audibility of cabinet resonance. If you were going to spend an extra $300 making the cabinet more inert than it had to be to be completely inaudible, then you are taking that money away from somewhere else. So in any area, be it how much bracing the cabinet has or which capacitors we use, each of these things is looked at carefully for making sense. But, especially at the beginning of a speaker design, the inception, there's a lot more than running test equipment since you are conceptualizing. Say we want to make something that costs $2,000. Does that mean that we make a little tiny two-way speaker? Is that giving people the best sound for $2,000? Or will we make a 3-way floor-standing speaker? And if we did that would we be compromising on the drivers?

RANADA: A lot of people don't understand how speaker designers work. They think they are like some sort of composer who thinks up a speaker and then builds it. But you are looking at a particular price point in the market, aren't you?
VOECKS: It is true that each model is thought of in a price range and that our goal is to make it clearly the best-sounding speaker in that price range. That's one of the things that doing a lot of double-blind listening up in Ottawa has shown to be of great value. We like to bring in competitor speakers at twice the price and make sure that under double-blind conditions everyone liked ours more. That gives us some breathing room to know that we are competitive and will be for the life of that model.
RANADA: How does one redesign an old speaker? When do you know to look at the design of an old model?
VOECKS: When someone else has come along with something that's strong competition. Or let's say that the quality of a driver improves substantially, allowing you to make that improvement. We don't make any of our own drivers, ironically, because we would then have difficultly meeting our standards for uniformity. If you build them yourself and you put two extra turns [of wire] on the voice coil, it's going to be out of spec. If you built it yourself, chances are that you are going to want to use it. Some people take the approach of using similar drivers across a broad price range, and the good ones will go in the more expensive speakers. I don't think that's acceptable because if $465 is a lot of money for somebody to spend, then they ought to have speakers that are every bit as good as the $465 speakers that were reviewed. They shouldn't be "sort of similar." Another approach is that you sell reject drivers to hobbyists. But I don't feel like getting into that. The truth is we don't have the Not Invented Here syndrome. There are people at driver manufacturers who are specialists, and they do a great job. We don't have any particular allegiance to any particular driver manufacturer; every model is different. There may be fifty drivers for one speaker that have to be auditioned to determine which is the best.
RANADA: Where do you think there needs to be more progress in audio? What kind of sound are you looking for when designing?
VOECKS: Accuracy. The ideal goal is: close your eyes and you can't tell whether you're in Symphony Hall or not. And, as you know, we are a long way from that.
RANADA: Where do you think the most progress will be made in the next few years?
VOECKS: At both ends of the system. Shockingly crude microphones are still being used that have resonances and high-frequency rises. That's just completely unacceptable. Typical mike technique, the million-mike method that became popular with multitrack tape recorders, is simply no way to make a recording. Fortunately the specialist companies like Reference Recordings and Telarc and Sheffield are probably having a nice impact on the bigger companies. The microphones will improve and they need to; the way they pick up instruments will improve; and speakers and they way they interact with the room can stand pretty drastic improvement.

"Shockingly crude microphones are still being used that have resonances and high-frequency rises. That's just completely unacceptable....the million-mike method...is...no way to make a recording."

RANADA: Would you see that as one of the major barriers to your goal, these interactions with the room?
VOECKS: Absolutely. It is extremely difficult to get audiophiles to pay as much attention to their rooms as they do to the number of widgets their preamp has. It's very frustrating, and it's understandable. It happens with audiophiles, salespeople, and reviewers. It's a lot more fun to plug in a new, neat, shiny component and hear, or imagine, some exciting improvement than it is to go to huge amounts of trouble measuring your room, or putting huge amounts of Sonex on the walls, or building in bass traps. I tell a lot of people that they ought to hire an acoustician if they have the kind of money they say they have. [They'll] get themselves a more neutral room.
RANADA: I've always told people, if you want a new audio system for nothing just move your speakers by a fool or two in any direction, and it will sound different enough to be like a new audio system.
VOECKS: That's absolutely true. We've tried to help out by having a room analysis program that gives you some idea of what shape your room is in at low frequencies and gives you some very simple guidelines for placing the speakers and the listening position.
RANADA: Do you think audio systems will be ultimately perfectible? Can we ever create a perfect audio system?
VOECKS: I don't think so, particularly because our standards will change as the systems improve. You remember there were tests with Victrolas in which no one could hear the difference, and there were similar tests about 20 years ago. *[See Roy Allison interview.—DR]* I find, if we do a simple live/miked test, that *no* speaker sounds right at all. They all sound wrong to varying degrees. But I think when we get more channels in the system—that's certainly a must—when we learn more about properly picking up the signal to begin with, I hope there will be progress.
RANADA: Do you think high-end magazines have helped or hurt the industry?
VOECKS: I think they have helped a lot in many respects. They've kept up people's interest in the field, and people live vicariously through the magazines. I think that's great and healthy for everybody. They can see what's going on and enjoy it vicariously, and when it comes time to buy a component they will at least have some familiarity with what's being said or being talked about. I hope that readers wouldn't blindly follow what anybody else says but instead do listening for themselves. That's awfully important.

RANADA: A lot of high-end editors dismiss double-blind tests out of hand and would never even consider submitting themselves to one.
VOECKS: For that viewpoint there is no good argument against them [the tests]. But it has just been so obvious in my experience with lots of these tests that all of us, even those who believe in double-blind tests, are very swayed by knowing what component is what. When we do it with speakers in the design process, I've brutally trashed exactly what I've been working on. It's interesting that when done with electronic components these tests tend to minimize the differences, or more correctly, they tend to expose just how minimal the differences are. In a speaker test you find night-and-day differences between most of them, so you end up being really rough on all of them.

RANADA: So you recommend double-blind testing to all those who have access to it.
VOECKS: Absolutely. Even a single-blind test is better than nothing at all.
RANADA: And that's where the listener doesn't know what is being switched.
VOECKS: Right.
RANADA: And a double-blind is when the switcher and the listener both don't know.
VOECKS: Right. What you do is have a third party go in and set up the system. Then they disguise it from you (make it invisible). Then you go in and switch at will.
RANADA: You've gotten into trouble, at least in some quarters, with your non-dismissal of what would in those quarters be called mid-fi equipment.
VOECKS: It's just not true that it has to be expensive to be good. And I think that one of the things that people forget about is having some perspective on the amount of difference between components. For instance, 1 constantly hear, "these two inter-

connects make a night-and-day difference in my system." Well clearly that's ridiculous because moving the speakers a foot would make a much bigger difference, let alone treating the room or choosing better speakers to begin with. So the perspective somehow gets lost.
RANADA: You use mid-fi components in your design process, don't you?
VOECKS: Yes. For instance, in the design process we use a lot of NAD equipment. It's always considered a very good value but it certainly would be looked down upon by hard-core audiophiles who spend megabucks. The truth is, the differences are straightforward and predictable between one electronic component and another. We've done one demonstration that came close to backfiring because we made it appear that we were playing large speakers and that we had really state-of-the-art [equipment] as the associated components, when actually we were playing our largest bookshelf speakers [Type K] and we were using the smallest receiver and cheapest CD player we could get. And people were really astonished. It was just a little bit of a demonstration to show that you should really sink the money in the speakers because that's what makes the biggest sonic difference. That's the component that needs the most money spent on it to get the best results. Also, when people have have preconceived notions, things sound different. If the amplifier is gigantic and weighs a hundred pounds and is extremely expensive, it will sound better to everybody than an amplifier that performed identically but looked like cheesy junk. So I certainly believe that double-blind tests are completely essential. I don't believe the results of any other kind of listening test.
RANADA: Beyond listening tests, how does a novice train himself to listen? It's something we more or less take for granted.
VOECKS: They certainly can be led astray very easily. The history of audio products is cluttered with speakers that grab your ears when you walk in the showroom. Certainly listening to live music and spending as much time listening to as many systems as possible will sharpen up your skills. Referring again to the NRC tests, it is nice to know at least that they will be going in the right direction. With more experience they will get there a bit more quickly and more surely.

* * *

4. Interview with Floyd Toole, Research Director

RANADA: How did you get into audio? Was it a childhood thing—like it was for some people?
TOOLE: I remember as a very young teenager putting 6- by 9-inch oval speakers in butter boxes thinking they sounded fantastic and filling them full of old cloth scraps and bits of household stuff. So I was into loudspeaker enclosure design at a very early age. I think the earliest significant enclosure that I built was a Karlson, the famous Karlson enclosure—now this would be in the mid '50s. I still have a pair of those from that period.
RANADA: How do they sound?
TOOLE: They sound dreadful. I took one into the NRC lab and measured it as it was intended to function. Then I took the loudspeaker [driver] out and put it in a test box and measured [it] so that I had some idea of what the loudspeaker itself was capable of. All I can say is that the loudspeaker enclosure is an acoustical meat-grinder—it'll make mincemeat out of anything you put into it. It's absolutely not the way to design a loudspeaker enclosure as far a timbral accuracy is concerned because, if you remember the Karlson, it was designed with the loudspeaker inside the box, and the front of it was firing up into a highly reflective cavity with a slot of [exponentially] varying width through which the sound was to emanate. The theory was that the short wavelengths would all concentrate at the narrow part and radiate omnidirectionally, and the long wavelengths would move down the slot and radiate omnidirectionally.
RANADA: But they don't do that.
TOOLE: No, they don't do that. And it also totally ignored the cavity resonances inside this totally undamped enclosure, which were horrendous.

"It was my intention to extend my sound localization investigations from... headphone [s] ...into real sources of sound out there in space, only to find [available] loudspeakers... almost universally bad..."

RANADA: So what does this experience tell you about authorities in audio?
TOOLE: It didn't tell me anything back then, although I have to say that even though it was the enclosure with the greatest pedigree that I built in that period of time, it was not the one I most enjoyed listening to. So I guess nature took its course, and there was by virtue of natural selection a tendency on my part to have good judgment although I didn't realize it at the time.
RANADA: Did you sign on at the NRC as an audio person, or did you just fall into audio?
TOOLE: I fell into audio professionally. I was studying electrical engineering and I got a postgraduate scholarship to study wherever I chose. I chose England because it was a more exotic country than our neighbor to the south. I don't regret that decision because spending five years in a European culture was formative for me. But when I did get there I was not able to pursue the line of study I had hoped to, which was transistor circuit design (this was 1960 when the transistor was a new electronic device). I was forced to change into a new line of investigation for a Ph.D. thesis project and I fell into a long discussion with Professor Colin Cherry, who was then and still is highly respected for his work in communications theory. As a personal sideline, he had a personal interest in sound localization and had done some work with Leakey on the localization of stereophonic images and the perceptual process that was involved. I just became intrigued with this—that you could apply engineering techniques to furthering our understanding of the perceptual process. I thought that this was a most marvelous field of investigation because there are some really tangible rewards if you learn how the human works. All sorts of new products may be possible. One thing led to another, and all of it led to me working with Bruce Sayers, who was a student of Cherry's. The project was on the interaction of visual and auditory information, to look at how we perceive visual space and how we correlate that with the auditory component of the total experience of walking around in our world. That's how the project started, and it went along in that direction for a short period of time. For the auditory component we used headphones and simple signals, presented through headphones, that you could move from the left ear through the head to the right ear—lateralization it is called. It became very clear before long that this in itself was an absolutely fascinating perceptual phenomenon. It was not at all well understood back then. There were still some pretty wild theories about how perceptual localization worked. I left the visual component of my investigations and concentrated on auditory localization exclusively for the rest of the project. In the end I learned some useful things I'm pleased to see are currently found in textbooks. So it makes me feel good to look back on that. When I came subsequently to join the National Research Council as a full-time employee, I had access to a good anechoic chamber for the first time. It was my intention to extend my sound localization investigations from the headphone situation into real sources of sound out there in space, only to find that loudspeakers that were available for use were almost universally bad, in technical terms. Bear in mind that we were listening in anechoic spaces to technical signals. One has to specify with some accuracy a waveform that you hope to deliver to the listener's ears. If the loudspeaker cannot transduce this electrical signal into an acoustical version, the experiment doesn't work. So my very first task turned out to be finding a loudspeaker that would do the job. There were several loudspeakers in the lab at the time, and none of them was very good. It turned out that the best of the loudspeakers, the one that seemed to exhibit the most potential, was one that I had brought back with me from England as a personal possession. It was a woofer and a tweeter from a then new company called KEF. I had, while I was there [in England] as a student, gone to stereo shows and listened to products. There were several very good products, and my final choice was dictated in part by budget, which after many years living as a student was extremely limited. I brought back the components of a system which I thought would have the potential of sounding decently good. I used these components to build a system that actually did sound significantly better than the com-

mercial products in the anechoic chamber. Eventually there is the temptation to say, "Since this works so well on one axis in an anechoic listening situation, I wonder how it sounds in a normal room." The first time I took this [speaker] out into a normal room began the long trek that has gone on for 26 years: to track down and try to understand the various phenomena, physical and perceptual, that are involved in translating measurements on a single axis in an anechoic room—or measurements on any axes in an anechoic room—to what we actually hear in listening rooms. It is a problem of sufficient complexity that it has occupied me much of my professional carrier over those years, and there is still lots left to be done.

RANADA: Could you capsulize the most important results of your work? Other people have talked about your work, and it would be interesting to compare what you think is most important to what they've found valuable.

TOOLE: It appeared clear to me that the final arbiter of sound quality is two ears and the brain—a human listener—and that much of the variability in assessments of real products lay in that domain. Having a substantial background by that time in subjective experimentation it was abundantly clear that not all people were born equal as listeners. In order to get consistent opinions from those who were even good listeners, you had to design an experiment so that they could focus their attentions on the aspects that you wanted them to comment on [and so they wouldn't] be distracted by less relevant factors. The notion of a controlled experiment as a component of a listening test was already well-founded in my mind and had been demonstrated to be useful over the years. I simply designed some listening tests using that background; I didn't invent anything. I just did the logical things, drew a curtain across the room, adjusted the speakers to equal loudness, and tested people over a period of several weeks—individuals to see how consistent they were and a number of different people to see whether there were any patterns in their preferences or opinions. We tried a scale of ten just because it was convenient. What was surprising was how simple it was to get relatively consistent opinions from quite large numbers of people. There were aberrant people with judgments that differed—aberrant viewed in comparison to the group)—[but] most people most of the time liked the same loudspeakers. This could not be accidental. Then the next logical step was to see if there was any consistent relationship to the measured performances [of the loudspeakers]. This is all history now, of course, but back then there were as many different opinions on how to measure loudspeakers as on how to design them [and] as there were designers. I just took the scattergun approach and said I don't know what the right answer is, and apparently neither does anybody else—if they do, they don't stand out from the crowd. Let's just try them all, and in the end perhaps there will be a pattern that will reveal itself. There was. The first clear-cut relationship was that most people most of the time preferred to listen to loudspeakers that exhibited frequency responses that were relatively flat and directivity patterns that were relatively uniform and distortion that was relatively low. All of those wonderful "motherhood" things that we as engineers were brought up to believe were correct! That went along for a period, with some streamlining of the listening test techniques, mainly in respect to the program material. Listeners are a variable in such experiments, but so is the program material. Program material in a listening test is equivalent to the test signal in a technical measurement. In a technical measurement, if you don't have the right test signal, then certain problems will not be revealed. The same is true of the subjective version of a measurement. If the music doesn't have the right spectral content and dynamic range or level, then certain real problems will simply not be heard. So one must be careful to choose material that has the potential of revealing the problems, otherwise you'll simply sit there being entertained. If you are in the business of doing listening tests, they are very time-consuming and boring operations. They just go on forever if you are not careful. Over the years we came to identify certain kinds of combinations of musical instruments and voices which appeared to be useful in revealing different

"...most people...preferred to listen to loudspeakers [with] frequency responses that were relatively flat and directivity patterns that were relatively uniform and distortion that was relatively low."

aspects of loudspeaker performance. And using these, rather than just what happens to be on the "charts" at the moment, proved to increase the efficiency of the listening tests. People came to their opinions more quickly and were more stable in their judgments. And people who took part in the listening tests over long periods of time came to know the program material. In fact, after a while you just stopped listening to it as music and you listened to it as a kind of device through which you can hear things. Eventually, once you go through the technical assessments, you sit back and relax and pick some music that you enjoy just to see whether your analytical assessments in fact had any merit. I must say, over the years we have been satisfied that the results of these rather controlled analytical tests were indeed very good predictors of one's ability to be satisfied in relaxed listening situations. And again it all makes sense. If, for example, in order to satisfy some listeners we had to have a 10 dB peak at 3 kHz and a 4 dB hump at 50 Hz—boom and tizz—then I think we might have been suspicious that something was seriously wrong. But that didn't happen. We found instead that smooth and flat and lack of coloration and good spectral balance and wide bandwidth were preferred by listeners. As loudspeakers over the years have gotten better at achieving these "idealized" technical objectives, so also have their ratings in our subjective tests improved. Now this was true—still is true—for a particular room in which we were doing these tests. This room has achieved some modest fame by having been accepted as the prototype for the recommended IEC listening room. It's not a perfect room; it's just a good room and in acoustical terms typical of a room that many of us might have as a listening room at home. But the reality of loudspeakers is that once they leave the factory door, neither the manufacturer nor anybody else knows where they will end up. The customer has a problem because when the customer listens to loudspeakers in the store it's highly unlikely that the listening environment resembles in any important way the kind of environment they have at home. If it does, they're very fortunate. The next step in this logical progression of investigation was to explore how important a room was. Rather, to explore what aspects of room acoustics—of the interactions between loudspeaker performance and room acoustics—are important in what we perceive. This is what has been the focus of our efforts in recent time. So it started more or less accidentally but it has progressed, I have to say, in a most monotonously logical way.

RANADA: Where do you see progress being made in audio? From what you have just told me, I suppose it would be loudspeaker/room interaction?

TOOLE: Well, I think that is the present frontier. The interaction of the loudspeaker and the room can be separated with some success into two domains—what happens below 300 or 400 Hz, where probably the room resonances are the dominant factor, and above 300 or 400 Hz, where perhaps the loudspeaker's frequency-dependent directivity and the proximity, and acoustical construction, of the room boundaries will become the dominant factor. At these higher frequencies—this is where stereo imaging and timbral signatures come together—one would logically expect there to be quite a strong interaction with the program material. If you know anything about the recording industry, you know that there are very, very wide variations in philosophy on how to mike a musical event. They range from a coincident stereo pair right through to a panorama of 30 microphones or more distributed throughout an orchestra. And somewhere in this comes the pop-music technique of multitracking. Since there is no standard technique for encoding the stereo signal, there can be no single perfect technique for decoding this stereo signal because the decoding process involves the loudspeaker directivity and the room—its shape and construction—and how the loudspeaker and the listener are deployed in the acoustical space, and finally, and not insignificantly, the listener's expectations. Different listeners have different objectives. I think that is a matter of simple fact. Some people really love the spacious, all-enveloping concert hall experience, and other people prefer a dry, analytical, highly focused kind of localization experience. Perhaps never the twain shall meet—because these are really poles apart.

RANADA: Shouldn't the equipment be able to deliver both?

TOOLE: Well, yes, that is true. Even now, we can do this. Eventually it will be done more effectively than we can do it now. In fact, in my own home system it was not possible to build loudspeakers with variable directivity, and for purely practical and aesthetic reasons it was not possible to build a room that had adjustable acoustics that one would wish to live in. One has to use spatial enhancement, and I have had that in one of my systems for several years. I use it because I find that to satisfy my mood and expectations I can achieve my ends more effectively by adding to the signal. But at the same time I am sufficiently a purist that if I am listening to recordings with natural acoustics—good recordings of classical music—then the ancillary apparatus is off. It's just me and two loudspeakers. The only embellishment I have found satisfactory is a large, acoustically good listening room, and nothing else. That has provided for me for several years the most marvelous listening experiences. While I embrace the world of technology and synthesis and add-ons and so on, I think the most important rule that must be applied to any of these devices is that there should be the means of turning them up, down, and off, according to one's will.

RANADA: Is it possible, even in theory, to create a perfect recording system?

TOOLE: It's possible but probably not in my lifetime. One approach would be to arbitrarily decide on a standard recording technique. This could be done and, based on that standard recording technique, one could evolve the optimum playback configuration. But that is an unrealizable ideal because I don't think the recording industry would be likely to go along with it.

RANADA: Is it possible to develop an ideal recording technique that would be able to do anything that you'd want to do with an audio system?

TOOLE: I don't know. I really don't know. My suggestion of picking one technique was not necessarily that it would be an ideal technique. It would be an arbitrary choice and arbitrary standard. The risk is that one will have chosen problems that will eventually be regretted. For the time being I am quite happy to live with the system the way it is, perhaps because it adds another dimension of interest. When one puts a record on, one decides quickly whether one likes it or not, but not everybody goes the next step, which I do, which is to try to unscramble the variables and to decide whether the reason I am dissatisfied with what I hear is because of the way the recording was made or because of what's happening in my listening room.

RANADA: Have you been able to figure this out with some degree of success?

TOOLE: Only on a very few occasions, mainly because of time constraints and a lack of knowledge of what was actually done at the recording situation. The most revealing have been the few recordings that one encounters where there is some data on how the recording was made. Then you can fill in pieces of the puzzle and start to learn really what it is in the recording that generates a satisfactory or unsatisfactory aspect to what you're hearing at playback for your particular system. The difficultly for the industry is that a reviewer may find a certain loudspeaker design very satisfactory in terms of stereo imaging, in that it meets his expectations for the recordings he just happens to have played. This may or may not coincide with the taste of all customers. There is no perfect solution to that, I'm afraid.

RANADA: What should a perfect audio system do? What is a fundamental design goal for the perfect audio system?

TOOLE: It would allow me to hear what the artist created. Actually, the perfect audio system includes the recording process. There will be a disparity between those who believe that "what the artist creates" is what he [the listener] hears when he goes to the concert hall—that's one camp)—[and there are those who feel] that the final artist in a recording is the recording engineer and producer. And they will be—by virtue of the microphones they've chosen and how they've chosen to use them and the electronic processing, if any, they've decided to apply. These people will determine most of what we hear when we play a record at home. I would like some assurance that what I hear is what they've created, so that

"...I am sufficiently a purist that if I am listening to recordings with natural acoustics —good recordings of classical music—then the ancillary apparatus is off. It's just me and two loudspeakers."

there is some integrity. Otherwise one is in this uncertainty zone where you are never certain of what color of light to use to view a painting. Perhaps this particular painter liked to work under "northern daylight." It's that sort of situation. Unless one has north light to view paintings, one never really sees the painting as it was intended to be viewed. I think there is a parallel with the audio world that an audio recording is a work of art. Along with the musical work itself goes the establishment of a stereo panorama or a soundstage or what have you, all of this being dictated by technical factors—microphones and how they are used; signal processing, if it's used, and how it's used. These are decisions that are made by people, sitting behind consoles, listening to loudspeakers in a room. One can only experience that as it was created if one has a similar setup.

RANADA: Would you be willing to extend to the listener the right of final creation?

TOOLE: I think the listener has every right to create whatever he wants. It is a free society. But the ideal situation is that you would have a default condition. You could turn all your processing off and say: this is what the maker heard, and that satisfied him. But I don't agree with his taste, so I'm going to add a little of this and I'm going to continue the recording process. I'm going to sit behind my console now and I am going to add my chosen special effects to the recording. And that's fine. It makes audio a participatory sport, not a spectator sport. I'm in favor of that. In my experience, if you participate in something you learn about it. You learn what equalization does to sound, and if you have a chance to manipulate sounds with equalizers and time delays and artificial reverberations and multiple loudspeakers or what have you, one learns to identify the sonic signature of these particular acts. You begin to say, "Ah, I hear a little nasality when you push up the 2 kHz equalizer slider. I think I heard that in so-and-so's loudspeaker." And you go back, and perhaps you may be able to make a measurement of so-and-so's loudspeaker and you find, lo and behold, there is a little bump around 2 kHz. You say, "Eureka! I now have learned to subjectively identify something that has a measurable relationship." This is true with the processing as well. I think one can acutally learn from having the opportunity to adulterate recordings. In the end you may choose to turn it all off or you may decide that you have indeed improved the recording. Even though what I say is, in the purest sense, true—that one ought to be able to hear what the recording engineer heard; that's an unrealizable ideal—there is also the other camp which I alluded to earlier: that it is the concert-hall experience we're after. Two loudspeakers in a room cannot recreate the full concert-hall illusion. There just aren't enough sources of sound. It is possible sometimes, by augmenting the two-loudspeaker stereo system, it may be possible to arrive at a more satisfying facsimile of, or version of, the real thing.

RANADA: So you have no philosophical objection to the use of ambience devices?

TOOLE: No, I use them myself.

RANADA: Some of the high-end camp, ultra purists, are not willing to put anything into the signal chain.

TOOLE: I have no problem with that. I'm in that camp also because I can turn my auxiliary processor off. I think it is important to be able to do that so that one can in fact get back to the original. If one is always satisfied with the original one gets a little suspicious because that suggests that every recording is perfect by definition, and that is demonstrably not true. I have some recordings that musically are wonderfully satisfying but that acoustically leave something to be desired. I find my acoustical satisfaction improved if I extend the bandwidth a little bit or enhance the ambience a little bit. Other recordings require none of that; they have been wonderfully, tastefully done to begin with. Two-loudspeaker stereo in a good room is perfectly adequate.

RANADA: You do, of couse, now have a major recording engineer down the hall [John Eargle]...

TOOLE: Well, in my decision to join Harman International, it was a significant consideration. Here is a company that runs the gamut of the audio industry from PA in live concerts, to monitor loudspeakers in recording studios, to loudspeakers for the home and the car. It seems to hit all the

Hip Boots

Wading through the Mire of Misinformation in the Audio Press

Editor's Note: *This is a very fat and very crowded issue, allowing only a fractional "Hip Boots" column. I find it irresistible, however, to comment very briefly on two pieces of exquisitely detestable journalism.*

Harry Pearson's response (?) to criticism.

In the last issue, I took HP to task in this column for not grasping the elementary fact that a digital sound recording before it is decoded contains only data, nothing but numbers. His reaction in the June 1992 *Absolute Sound:*

"...I found myself (once again) attacked in The Audio Sleazoid by the master himself who ran on blah-blah-blah, head, as usual, stuck in an improbable, though no doubt familiar, position." No argument, no refutation, no facts, no retreat from his smug techno-illiteracies, just that depressingly witless, impotent little tantrum. That's an audio authority?

Robert Harley on music and power amplifiers.

There's an ancient joke about the double bass player who, on his day off, decides to go to the concert and listen to his own orchestra. Next day he says to another double bass player: "Hey, can you imagine, when we play oompah-pah, oompah-pah, do you know what the others are playing?" "What?" asks the other. "They're playing tra-la-la, tra-la-la!"

Robert Harley actually believes that's the way it is, as witnessed by his "Follow-Up" on the Boulder 500AE power amplifier in the April 1992 *Stereophile.* Disagreeing with his fellow staff member Lewis Lipnick on the "sound" of the 500AE, he points out that LL is the contrabassoonist of the National Symphony Orchestra and, since he sits in the middle of the orchestra, his judgment of the sound of live instruments is necessarily skewed. RH trusts his own judgment better when it comes to the "liquidity," the "rhythm and pace," etc., of a power amplifier and generally mistrusts any amp that wasn't designed by ear by a cultist without a degree.

It would follow, then, that Rostropovich can't be trusted to approve the sound of his own recordings because he conducts only a few feet from LL's chair. Even within the intellectual context of *Stereophile*, RH has become an embarrassment.

Preamplifiers

(continued from page 38)

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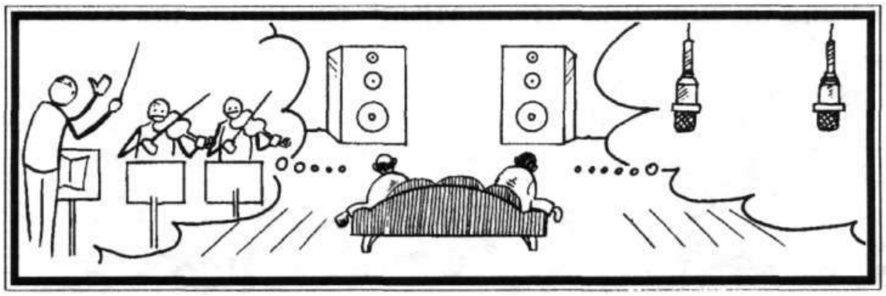
bases. It's an interesting possibility to see what this company can do to try to close that loop a little more: to bring monitor loudspeakers and hi-fi loudspeakers closer to together. And through our improved understanding of loudspeaker/room interactions to allow recording engineers to monitor more closely some of the things people at home will be listening to. To allow people at home to perhaps come closer to what recording studios are generating for them. •

Recorded Music



As will be immediately apparent to our readers, something quite new is being presented here: the music critic as "objectivist." Well, maybe not strictly in the equipment reviewing sense but close. David Ranada, former Technical Editor at Stereo Review *and* High Fidelity, *and now a Contributing Editor of this publication, is an audio expert with a degree in music and appears to fit the profile of the "rare bird" we could only speculate about in the last issue. Incidentally, the condensed tabular review format launched in that issue was well received and will be used again from time to time to catch up on our CD backlog.*

Mozart's *Die Zauberflöte:* a Compleat Review of Three Recent Releases

By David Ranada
Contributing Editor at Large

Last year, in case you hadn't noticed, was celebrated as the bicentennial of Mozart's death (on December 5, 1791). Among the flood of Mozart recordings released in commemoration were the three performances of Mozart's last opera, *Die Zauberflöte* or *The Magic Flute,* reviewed here. The first performance, conducted by Sir Charles Mackerras, is Telarc's first full-length opera recording. The second, from EMI/Angel, is the first original-instrument performance with dialogue and is conducted by noted original-instrument iconoclast Roger Norrington. The third recording, conducted by Sir Georg Solti, is the only stereo remake of the work (his first recording of the opera is also on London). For cast lists, CD information, and other particulars see page 65. This review is an unusually elaborate attempt to cover things not all normally mentioned in a single review of recorded music. I'll start, however, with what readers of this journal are perhaps most likely to be interested in.

Sound quality.

All the recordings have sound qualities representative of the standard "sound" for their labels. London's set, recorded in the Großer Saal of Vienna's Konzerthaus, has the most traditionally "hi-fi" recording quality. The dynamic range seems restricted, tastefully to be sure, as it is in most of the classical releases from PolyGram labels. Woodwind detail is sometimes lost in the string-oriented sound. The strings themselves are lush; that's simply how the Vienna Philharmonic sounds. Reverberation is of a medium-to-large hall, and the orchestral layout is standard (first and second violins left, cellos and violas right). As in his first recording of the work, Solti seems to use a celesta instead of the glockenspiel called for in the score. It sounds fine, even though the instrument wasn't even invented till the century after Mozart died. (First important use: Tchaikovsky's "Dance of the Sugar Plum Fairy.") The other two recordings use a glockenspiel; EMI's is harshly recorded.

The EMI effort is more obviously, but not obnoxiously, a multimike production. The unusual—but welcome—woodwind detail is the giveaway, with assistance from the close-in sound of the strings, the unnaturally prominent wooden transverse flutes, and the sometimes audible left-channel continuo fortepiano. The dynamic range is wider than on the Solti recording but not as wide as on the Mackerras. EMI used its Abbey Road Studio No.l as the recording venue, and it sounds appropriately smaller than the Konzerthaus hall used by London. Some may find the EMI sound quality a bit too up-front and dry. But this is also closer to the type of sound that Mozart would have encountered in the smallish theaters of his time and it also matches Norrington's conception of the work.

Included in the EMI libretto booklet is a diagram of Norrington's orchestral layout, which seeks to follow 18th-century seating plans as seen in various documents of the period. The first and second violins are divided left/right, the cellos and basses scattered around the stereo stage for more intimate sonic contact with the other instrumental groups. As heard here, the instrumental lines—particularly the woodwinds—are clearer here than in the other two recordings.

As might be expected, the Telarc recording has overall the most realistic sound and the widest dynamic range. It should be played at a higher volume setting for the most realistic results. The orchestra is at medium/far distance and, at times, this distance combined with the hall sound sometimes clouds the details, especially the woodwinds and the words of the chorus. Also, sometimes the miking gives the strings "first-desk" sound in which you can detect the vibrato of individual string players. This is undesirable, since the purpose of orchestral string vibrato is not to make the ensemble sound scrawny but to increase the richness of the sound. With a larger orchestra, as in the London recording, picking out individual players is more difficult. The rich-sounding but basically vibratoless string playing in Norrington's performance is not subject to this problem. On the whole, the string playing on the Telarc recording is not as razor sharp as it is on the two others. The first and second violins are divided left/right. Likewise, the violas and cellos are split left/right, an unusual arrangement that only rarely produces significant musical benefits in this work.

Sound effects.

Most listeners who know *Die Zauberflöte* only from previous recordings or the occasional live performance probably don't realize that Emanuel Schikaneder's libretto calls for a great many sound effects, especially thunder of various types, which are often omitted in recordings and performances. All the sound-effect cues I could find in the score and libretto are listed in Table III, along with one cue for speaking over the music (first on the list), the locations of the threefold B-flat chord which serves a mysterious function in Act II's initiation rituals, and the scene in which Tamino is to play his flute during the dialogue. As you can see, this list has been well served by these three recordings, any one of which contains more sound effects than earlier realizations.

Surprisingly, it is the musically most "conservative" of the performances—the Solti—which most fulfills the requirements of the libretto. Of particular interest are the charming twittering birds in "Holde Flöte" (Act I Finale) and the "crackling fire," "howling wind," "dull sound of thunder," and "rushing water" during the March of the Trial by Fire/Water (Act II, Scene 28). The latter effects are of particular symbolic import to any interpretation of the opera (remember the ancient elements of earth, fire, air, and water) and should have been included in the other two recordings. Of supreme symbolic significance in this regard is the threefold B-flat chord first heard in the middle of the overture. It returns at various times in the Act II in dialogue scenes related to the initiation of Tamino and Papageno. Only Solti performs all eight repetitions of this passage called for in the libretto. (Mozart actually wrote down the music only once).

Only Norrington has his orchestra perform the various loud, isolated chords called for in the libretto: once before the Queen of the Night's first entrance, and twice in Act II, Scene 22. These chords are specified only in Schikaneder's libretto; they are not notated by Mozart in his score. But they easily could have been improvised during his rehearsals. I also like the way, in Act II, Scene 17, Tamino in the Norrington performance plays a ornament-stripped version of the flute melody he will play in the initiation rites. (Mozart supplies no music here; the libretto specifies only "Tamino blows on his flute.") The removal of the ornaments is obviously meant to be symbolic of Tamino's "incomplete" and unworthy nature prior to the initiation scene.

Album	**Telarc *CD-80302***	**EMI *CDS 7 54287 2***	**London *433 210-2***
Number of discs	2	2	2
Orchestra	Scottish Chamber Orchestra	London Classical Players	Vienna Philharmonic Orchestra
Conductor	Sir Charles Mackerras	Roger Norrington	Sir Georg Solti
Principal Cast Members			
Pamina	Barbara Hendricks	Dawn Upshaw	Ruth Ziesak
Tamino	Jerry Hadley	Anthony Rolfe Johnson	Uwe Heilmann
Papageno	Thomas Allen	Andreas Schmidt	Michael Kraus
Sarastro	Robert Lloyd	Cornelius Hauptmann	Kurt Moll
Queen of the Night	June Anderson	Beverly Hoch	Sumi Jo
Papagena	Ulrike Steinsky	Catherine Pierard	Lotte Leitner
Monostatos	Helmut Wildhaber	Guy de Mey	Heinz Zednik
Production			
Recording Date	July 13-22,1991	December 1990	May and December 1990
Venue	Usher Hall, Edinburgh	Abbey Road, London	Konzerthaus, Vienna
Producer	James Mallinson	David R. Murray	Michael Haas
Engineer	Jack Renner	Mike Clements	John Pellowe
SPARS Code	DDD	DDD	DDD

The Telarc thunderclaps are the most spectacular, as one might have expected. Connoisseurs of recorded thunder will enjoy the variety of natural cracks, crashes, and rolls. Some of the EMI thunder seems synthesized and/or assisted with a bass drum; London's thunder is hampered by the compressed sound quality. Both EMI and Telarc send in roaring lions in Act II, Scene 19, although no roaring is specified in the libretto. Telarc's leonine contributions (from the Cleveland Metroparks Zoo) sound a bit too loud—and too closely miked! EMI's more well-mannered beasts, Pagan, Mitzi, and Maisie, appeared through the courtesy of the Zoological Society of London.

Dialogue.

While thunderclaps may be spectacularly theatrical, they are not inherently dramatic when compared with the combination of spoken dialogue and music which are the core of the opera. If a listener is to fully understand the messages of *Die Zauberflöte,* the dialogue must be there along with the music. Many previous recordings such as Böhm's (Deutsche Grammophon), Jordan's (Erato), Karajan's (Deutsche Grammophon) tended to see these works as "grand opera" with comic interludes. This is reflected in their drastic abridgment of the dialogue. Harnoncourt—ever on a different path—replaces most of the dialogue with a narrator(!) who sounds as if she were reading, in German, to a bunch of kindergarten pupils. While this opera may at times have a childlike simplicity, it is definitely not for children. Klemperer's performance (EMI) goes so far as to omit the dialogue altogether. Far from showing respect for the composer, these abridgments severely distort Mozart's intentions—not to mention Schikaneder's as librettist—and disqualify Harnoncourt's and Klemperer's recordings for consideration as one's sole *Zauberflöte.*

Musicologist Jacques Chailley's important book about *Die Zauberflöte* more than adequately documents (see "Recommended reading") that mystical, even Masonic symbolism pervades *Die Zauberflöte* from the deepest levels of musical content to the most trivial details of the staging specified in Schikaneder's libretto. Mozart's music amplifies the atmosphere of a libretto that, in its complete form, exquisitely balances farce and an almost religious spiritual fervor. Removal of portions of dialogue upsets this balance, since this eliminates crucial information about the characters' personal histories, relationships, and motivations.

Fortunately, these three recordings contain some of the most complete performances of the dialogue yet available. Although I didn't make a count, I did mark up a libretto as to which recording includes what. In number of lines preserved, Telarc and EMI are about tied for first, followed by London, which preserves a little more than the traditional ratio. But only with the EMI recording do we learn from the dialogue that Tamino is twenty years old, or the circumstances behind Pamina's escape from Monostatos in Act I. Only from the Telarc do we hear how Pamina was originally abducted (while sitting in a grove of cypresses). And only in the London recording does Sarastro say (Act II, Scene 1) that "Evil prejudice shall disappear, and it will disappear as soon as Tamino himself comes into the possession of the greatness of our difficult craft." (May the Force be with him!) All the recordings preserve more lines for Monostatos than usual, which is helpful in defining this enigmatic role.

However, because of cuts in the dialogue, from none of these recordings do we find out that Papageno's mother once served the Queen of the Night, that Tamino's ambition might be "to someday reign as a wise ruler," or that Pamina's father refused to give one of the symbols of his powers (the "sevenfold sun-circle") to the Queen of the Night and instead gave it to Sarastro, thereby provoking the Queen of the Night's vengeful coloratura. Perhaps this latter passage of dialogue (Act II, Scene 8) has been eliminated so as not to offend modern sensibilities. In it, the Queen of the Night quotes Pamina's father (the King of the Night?) as saying to her: "Your obligation is to surrender yourself and your daughter to the leadership of wise men." The irony of the libretto, which looks so antifeminist in such a passage, is that Pamina *is* ultimately initiated into the company of "wise men," albeit only as a partner of Tamino. The Telarc is the only one of these recordings to omit entire scenes (Scenes 9 and 10 of Act I, which help explain Monostatos's motivations). From all this it should be clear that an understanding of *Die Zauberflöte* will be greatly deepened by a reading of a complete and accurate translation of the libretto. Such a translation is easily obtained (see "Recommended reading" below), so you can easily make up all these omissions.

A few of the principals in Mackerras's and Norrington's recordings seem somewhat uncomfortable speaking (and, at times, singing) German, Mackerras's singers very much more so than Norrington's. Perhaps this unease accounts for the stilted, overconversational, and untheatrical delivery of the dialogue throughout much of the Mackerras recording, especially Act II. This is by far the most serious fault of the Telarc performance and is one of the two traits of the recording which bring it down from the top rank that it would otherwise occupy. One specific example, although not strictly speaking a dialogue passage, will suffice. In the middle of Scene 29 of Act II, Papageno, while seeking Papagena, counts off "one, two, three" out loud, blowing his panpipe between each number. In the theater it is a thoroughly charming moment for a Papageno who is an effective actor. With an audio recording, you only have the musical/dramatic context and your imagination to get you through. As Table II shows, Mackerras's recording takes 35 seconds to get through this passage, a full nine seconds longer than either Norrington or Solti. In context, it seems an interminable length of time. After waiting this long for a Papagena to show up, any audience would be like Papageno: ready to commit suicide.

Pacing of the dialogue in the Norrington recording is a rapid as that of his conducting. There is a feel of real character-to-character interaction—of acting—rather than singers simply waiting to get their next cue as in the Mackerras performance. But there were times when, although the de-

Table I: Overall Timings of Musical Numbers

Musical Number	Recordings reviewed here: Mackerras *(Telarc)*	Norrington *(EMI)*	Solti *(London)*	Other recordings: Böhm *(DG)*	Klemperer *(EMI)*	Harnoncourt *(Teldec)*
Overture	6:30	5:56	6:28	7:10	7:13	6:24
First Act						
1. Zu Hilfe!	6:02	5:44	6:08	6:43	6:56	6:33
2. Der Vogelfänger	2:32	2:17	2:27	2:34	2:42	2:58
3. Dies Bildnis	3:40	3:23	3:53	4:35	4:11	4:00
4. O zittre nicht	4:39	4:10	4:35	5:21	5:10	4:29
5. Hm!Hm!Hm!	5:50	5:38	5:52	6:26	6:46	6:32
6. Du feines Täubchen	1:38	1:40	1:40	1:44	1:56	1:36
7. Bei Männern	2:50	2:35	2:47	3:15	3:36	4:14
8. Finale	22:52	20:54	23:31	25:53	24:50	24:46
Second Act						
9. Marcia	2:35	2:36	2:24	*1:22	4:07	2:40
10. O Isis und Osiris	2:32	2:12	3:06	2:54	3:16	2:50
11. Bewahret euch	0:58	0:43	0:52	0:55	1:10	0:52
12. Wie? Wie? Wie?	2:53	2:53	3:01	3:37	3:27	3:05
13. Alles fühlt	1:14	1:17	1:13	1:17	1:23	1:13
14. Der Hölle Rache	2:54	2:45	2:49	2:55	3:13	2:58
15. In diesen heil'gen Hallen	3:44	2:43	3:44	4:02	4:49	2:58
16. Seid uns zum zweiten Mal	1:31	1:24	1:27	1:47	1:56	1:22
17. Ach ich fühl's	2:58	2:51	3:53	4:27	4:09	4:00
18. O Isis, und Osiris	2:44	1:51	2:55	3:06	3:13	2:53
19. Soil ich dich, Teurer	2:44	2:45	2:52	3:01	3:17	3:35
20. Ein Mädchen oder Weibchen	3:57	3:25	3:43	3:54	4:19	4:34
21. Finale	29:49	26:12	28:56	32:28	32:28	31:42
Music, Total	117:06	105:40	118:16	129:26	134:07	126:14
Recording, Total	148:06	137:51	151:26	147:21	134:07	143:40
Dialogue and Pauses, Total	31:00	32:11	33:10	17:55	0:00	17:26
Dialogue as % of Total	21%	23%	22%	12%	0%	12%

*No repeats taken in Böhm recording. Both repeats taken in others.

Table II: Timings for the Finale of the Second Act

Bars	Tempo Marking	Cue	Mackerras	Norrington	Solti
1- 93	Andante	Bald prangt	3:39	3:19	3:44
94-189	Allegro	Sollte dies dein Jüngling sehen	2:03	1:56	2:10
190-248	Adagio	Der, welcher wandert	3:20	2:22	3:19
249-277	Allegretto	Was hör ich?	1:20	1:04	1:18
278-361	Andante	Tamino mein!	3:36	2:58	3:12
362-388	Marcia-Adagio	Wir wandelten	2:22	2:02	2:31
389-412	Allegro	Triumph! Triumph!	0:42	0:46	0:44
413-529	Allegro	Papagena! Papagena!	2:36	2:21	2:25
530-533	(Spoken)	Eins! Zwei! Drei!	0:35	0:26	0:26
534-542	Andante	Nun wohlan, es bleibt dabei	0:36	0:26	0:24
543-575	Allegretto	Halt ein, o Papageno!	0:53	0:50	0:52
576-615	Allegro	Klinget, Glöckchen, klinget	0:44	0:40	0:39
616-744	Same tempo	Pa—pa—pa—pa—pa—pa	3:06	2:21	2:19
745-814	Piu moderato	Nur stille!	2:01	1:56	1:49
815-823	Same tempo	(Scene change)	0:23	0:17	0:22
824-827	Recitative	Die Strahlen der Sonne	0:20	0:15	0:18
828-845	Andante	Heil sei euch Geweihten!	1:02	0:59	1:07
846-920	Allegro	Es siegte die Starke	1:13	1:12	1:14

Table III: Extramusical Cues

	Mackerras (Telarc)	Norrington (EMI)	Solti (London)
First Act			
Scene 1 Dialogue—Tamino speaks his final words while the orchestra is playing the introduction to "Ein Vogelfänger"	Yes	Yes	**No**
Scene 2 Dialogue—[1]"[Papageno], who has played on his pipes several times up to here"	No	Yes	No
Scene 5 Dialogue—Before Tamino's "Was ist das?": "Heavy, shuddering chord" (no thunder)	Thunder	Chord & thunder	Thunder & wind
Scene 5 Dialogue—-Then three thunderclaps alternating with three "Sie kommt!"s (no exit thunder)	Only 2, with exit thunder	3, with exit thunder	3, with wind
Scene 75—Tamino: "Wie stark ist nicht dein Zauberton" "Birds twitter while he is playing"	No	No	Yes
Second Act			
Scene 1 Dialogue—Three repetitions of the threefold B-flat major chord	Only 1	3	3
Scene 2 Dialogue—At start: "Thunder rumbling in the distance "	Yes	Yes	Yes
Scene 2 Dialogue—Thunderclap after Papageno's "wollf ich dir's schon sagen aber so—"	Yes	Yes	Yes
Scene 2 Dialogue—"Strong thunderclap" after Papageno's "eiskalt läuft's mir über den Rücken"	Yes	Line omitted	Line omitted
Scene 2 Dialogue—"Very strong thunderclap" after Papageno's "Ich wollf, ich wär' ein Mädchen!"	Yes	Yes	Yes
Scene 5, No. 12—("Wie, wie, wie?") "A terrifying chord from all instruments, thunder, lightning and crashing, at the same time two strong thunderclaps" at "Hinab mit den Weibern"	Thunder	Thunder	Thunder
Scene 5 to 6 Transition—threefold chord once	Yes	Yes	Yes
Scene 8—Thunder accompanies the Queen of the Night's entrance. Exit thunderclap is not in libretto.	Yes, with exit thunder	Yes, with exit thunder	Yes, but no exit thunder
Scene 15 Dialogue—Thunderclap after "Ich heiße—"	Yes	Yes	Yes
Scene 17 Dialogue—Tamino plays flute	Yes	Yes	Yes
Scene 19 Dialogue—threefold chord once	Yes	Yes	Yes
Scene 19 Dialogue—"The lions come out" (no sound specified in the libretto)	Roar	Roar	Nothing
Scene 19 Dialogue—threefold chord twice	Only 1	2	2
Scene 22—-"Zurück!", then thunderclap, then "Zurück!", then another thunderclap, both accompanied by "a loud chord"	Thunder only	With chords & fire	Thunder only
Scene 28 Beginning—-"A waterfall is heard to rush and roar"	No	No	No
Scene 28 March—-"Crackling fire, howling wind, dull sound of thunder, and rushing water"	No	No	Wind & thunder only
Scene 30—At the Three Ladies' "Ihr Kind soll deine Gattin sein" "one hears dull thunder and rushing water"	Yes	Yes	Thunder & wind
Scene 30—Before "Zerschmettert, vernichtet is unsere Macht" "a very loud chord, thunder, lightning, storm"	Thunder & wind	Thunder only	Thunder & wind

Table IV: Error Correction Data

	Mackerras (Telarc)		**Norrington (EMI)**		**Solti (London)**	
Disc Characteristic	***Disc 1***	***Disc 2***	***Disc 1***	***Disc 2***	***Disc 1***	***Disc 2***
Overall playing time	74:11	77:44	72:37	66:04	77:24	74:22
Total number of block errors	13,710	19,335	5,338	16,171	64,528	64,702
Average block errors per second	3.08	4.15	1.23	4.08	13.89	14.50
Number of interpolations	0	0	0	0	0	0
Pressed by	DADC		Sonopress		PDO, USA	

livery was sufficiently rapid, I thought the singers were not having enough fun with their roles. Too often, in their inflections and vocal coloration, things were a bit too serious. Perhaps Norrington has slid too far towards a symbolic interpretation of *Die Zauberflöte,* in which everything, even the funny bits, has a serious, hidden meaning.

The most idiomatic-sounding dialogue is provided, not unexpectedly, by the mostly German-speaking cast of Solti's recording. While not as rapid-fire as on EMI, the delivery here is also not nearly as stilted as in the Telarc recording. It's too bad that not more of it was included. My main complaint here is that the London recording's dialogue, like the Telarc's, is at points delivered in too conversational a tone, as if the listener were eavesdropping on normal-voice conversations. This style of dialogue is a throwback to the days when actual actors did the speaking parts for the singers in an opera recording (such seems to have been the case in Böhm's recording). The spoken dialogue then had the too-intimate atmosphere of a German radio play. It's not that bad here, but it does get close, as in Tamino's first spoken passage and Sarastro's under-delivered conversation with his acolytes in Act II, Scene 1.

Conducting.

Norrington's is the most thoroughly thought-out original-instrument CD performance of a Mozart opera I have yet heard (and I've heard them all, except for John Eliot Gardiner's *Idomeneo* and *La Clemenza di Tito* on Archiv). Norrington's pacing is that of a Broadway musical, which *Die Zauberflöte* resembles in very many ways: nothing holds back the flow. My ears quickly adjusted. The drama moves along crisply and theatrically, aided by the sparkling delivery of the dialogue, yet Norrington takes time, relatively speaking, where warranted. For example, the Papageno/Pamina duet "Bei Männern," while timing out as fastest in Norrington's performance, does not sound at all rapid in context. He also preserves other important moments of musical repose and solemnity (the opening of Act II and Pamina's "Tamino mein...," and the chorus's climactic "Heil sei euch Geweihten" near the end).

While traditionalists might balk at the rapidity of Pamina's "Ach, ich fühl's" and of Norrington's nearly in-tempo scene-change music after the final destruction of the Queen of the Night (Mozart marks no tempo change, though all conductors slow down to some degree), the only section I thought could use less of a push was the Act II chorus "O Isis und Osiris." Certainly, of all the live and recorded performances I have heard—including the other two reviewed here—Norrington's is the only one to have invested Tamino's long accompanied recitative at the beginning of the Act I Finale ("Die Weisheitslehre") with a combination of musical and dramatic animation preventing it from seeming as interminable as a monologue from *Parsifal* (now *that's* a work that really needs revisionist tempi!). Finally, those who think that Norrington is only after pedantic "authenticity" should hear the positively Wagnerian tempo variations in Tamino's "Dies Bildnis," which here is given with an extraordinary degree of passion.

Mackerras, like Norrington, chooses some of his faster-than-traditional tempi on the basis of quite sound historical-musicological considerations, which both conductors detail in their program notes. But Mackerras hasn't gone as far as Norrington in speeding everything up; he has quickened some of the slow sections but only rarely speeded up the fast sections—the dynamic range of tempos is therefore compressed, restricting the variety of dramatic pacing that can be achieved. And, as Table I shows, even after his increases in tempo, Mackerras is often slower than Solti! It is this lack of variety and dramatic drive that causes my dissatisfaction with Mackerras's pacing, which is the second major disappointment of the set. In crucial moments he ends up lowering the dramatic temperature rather than raising it. For all the faster tempi, it is a curiously low-voltage performance, and a far cry from Klemperer's and Harnoncourt's much slower but extremely intense renditions.

A prime example are the tempos he employs in the Act II Finale, the longest continuous stretch of music in the work. Table II is a breakdown of timings for each marked change of tempo in the Finale (as well as the transition to the final tableau, which has no tempo change indicated from the previous section). Mackerras is not only the slowest of these performances in overall timing of the Finale (see Table I), but Table II shows that he is the slowest for more than half of the different subsections of the Finale. And those sections where he is the slowest serve to drag down the emotional level when it should be peaking, since they are the sections containing the denouements of the three subplots of the opera: 1. the transcendent Tamino/Pamina-reunited duet in bars 278-361 ("Tamino mein!"), 2. the extended scene with Papageno and then with Papagena in bars 413-744 ("Papagena!"), and 3. the final destruction of the Queen of the Night in bars 745-814 ("Nur stille, stille, stille"). The end result of this dawdling is to vitiate the musical climax of the work: the final return to the home key of E-flat major with the slow choral release of "Heil sei euch Geweihten!" ("Hail to you initiates! You have penetrated the darkness."). Among other numbers in Act II, Mackerras also drags out—to no musical, dramatic, or musicological purpose—what would otherwise be a highlight of any performance, Papageno's "Ein Mädchen oder Weibchen."

Mackerras is, however, the only one to provide a musical appendix to his performance: the Tamino/Papageno duet "Pamina, wo bist du." This was claimed by Schikaneder to have been written by Mozart and on CD is otherwise available only, I believe, on the Sawallisch recording (EMI). While of minor musical interest, it is nice to have around even if it may not be entirely Mozart's work.

Solti's performance is more traditional in choice of tempi in precisely those sections where Mackerras and Norrington are at their most radical. But so convincing are the latter conductors' musical and historical arguments for the use of quicker tempi that those passages in Solti's performance sound positively glacial by comparison, even though, overall, Solti is quite a bit more rapid than older performances like Böhm's and Klemperer's (see Table I) and is, as stated earlier, sometimes faster even than Mackerras. This sluggishness is particularly evident in "Herr, ich bin zwar Verbrecherin" (Act I Finale, Scene 18), "Ach ich fühl's," and Sarastro's "O Isis und Osiris." These exceptions aside, Solti's performance is extremely well paced, especially the sequence of tempo changes in the Act II Finale and the "Weisheitslehre" recitative in the Act I Finale.

All three conductors seem to have made their singers unusually aware of the stylistic niceties of 18th-century performance practice. There are more appoggiaturas and cadential flourishes than Mozart opera recordings usually receive, even in Solti's reading. Decades of musicological insistence seem to be having a positive effect here.

Singing.

Luckily—for the length of this review—these three recordings are blessed with extremely strong casts, so there is relatively little to gripe about here. The best-sung set all around is London's, with not a weak performance in the bunch either in vocal technique or characterization, and with outstanding work from the Pamina (Ruth Ziesak) and the Papageno (Michael Kraus). The entire cast also sounds as if they knew what they are singing about, something not always evident from the Telarc and EMI recordings.

The vocal quality is not so uniformly high in the other two albums. The weakest link in both sets is the Queen of the Night. Telarc's high-powered June Anderson has the irritating habit of hitting some of her notes slightly off pitch (too high, which is unusual). She gets the "hard" ones, like the high C's and F's just fine; it's the "easy" notes lower down she has trouble with. This fault gets more annoying the more one hears it. Less irritating is Beverly Hoch on EMI, who instead suffers from an excessive and inauthentic vibrato. It often clouds her otherwise pitch-accurate passage work.

Aside from the Queen, the cast of Norrington's set is fine except for the fact that their singing suffers from the same overseriousness that characterizes their delivery of the dialogue. The voices are "smaller" compared with those on London or Telarc, but they blend very well with the sonic quality of the period-instrument ensemble.

Barbara Hendricks, the Telarc Pamina, sings beautifully, but I'm bothered by her attempts to enunciate German. Her vocal timbre doesn't seem optimally matched to that language. But I rather liked the wide-eyed innocence Jerry Hadley brings to the role of Tamino in the Telarc set, although his pitch control suffers when pushing too hard. Monostatos is definitely underplayed on Telarc by Helmut Wildhaber. His Act II aria "Alles fühlt der Liebe Freuden" falls entirely flat.

Program Notes/Libretto.

In addition to the conductors' comments in the Telarc and EMI releases, the program-note/libretto booklets of all three new recordings have other things to recommend them. The Telarc recording features a very good (accurate) translation, by Susan Webb, of Schikaneder's libretto as performed on the recording. However, her translation of the absolutely complete text, including all scenic and stage directions, is also available in a handy volume of translations of Mozart's mature operas, listed in the References below. EMI/Angel's booklet has a comparatively short but very informative essay on the genesis of the opera by Peter Branscombe, one of the foremost musicological experts on the work. His monograph on *Die Zauberflöte* is also recommendable and is likewise listed at the end of this review. Branscombe also provided good translations of the dialogue portions of the Angel libretto. The translations of the musical portions are uncredited and identical to those provided with the Klemperer recording. London's booklet contains an essay on the opera by noted Classical-period musicologist H. C. Robbins-Landon, which goes into greater detail on the Masonic background of the opera than either of the other recordings' essays. *His* moving portrait of Mozart's last year, which included the composition of the *Die Zauberflöte*, is also on the recommended list. London's fine translation of the libretto is uncredited.

Presentation.

The EMI/Angel recording's box is the most relevant to the content of the opera, being covered with Egyptian motifs (which are important in Masonic symbolism as well as to the specified scenic design of the work). It also includes a portion of Schinkel's famous 1816 star-covered dome setting for the Queen of the Night's entrance in Act I, Scene 4. As a fascinating comparison, the London box is a photograph of a set for the same scene in a modern Viennese production (the Queen of the Night appears to travel around in a bubble, like Glinda in *The Wizard of Oz).* Telarc's artwork is a modern redrawing of the Schinkel design (by John Maggard). Both Telarc and EMI helpfully give separate track numbers to the musical pieces and the intervening dialogue; London provides cues only for the musical sections. Telarc also more finely subdivides the finales with track numbers, although none of the performances provide as many of these as I would have liked. Pressing quality was acceptable throughout. None of the discs caused digital interpolations—all the errors were completely eliminated by the CD error-correction system. (See Table IV for complete error-correction data).

Summary.

Although they are not all mentioned here, I have heard all the commercial stereo recordings of *Die Zauberflöte* presently available on CD except Marriner's (on Philips). If you must have only one *Zauberflöte* recording, I'd recommend either the new Solti, for a quickly paced yet traditional-sounding performance with very good singing, or the Norrington, for authentic tone colors and a palpable sense of theatricality and ensemble drama. Next in sequence I would place Telarc's effort—which is most valuable for the relative completeness of its dialogue, its sound quality, and its spectacular sound effects—alongside the old Solti

Nikolaus Hamoncourt's Beethoven Symphonies on Teldec

By David Ranada
Contributing Editor at Large

Ludwig van Beethoven: 9 Symphonies. The Chamber Orchestra of Europe, Nikolaus Harnoncourt, conductor. Teldec 2292-46452-2 (5 discs, DDD; recorded live June 29-July 5, 1990, and June 21, 1991; producer, Helmut Mühle; engineer, Michael Brammann; released 1992).

Today's conductors have split into two camps: the neo-Furtwänglers and the neo-Toscaninis. And then there's Nikolaus Harnoncourt, who seems to follow nobody's drummer but his own. Those of you familiar with recordings of baroque music will know of Hamoncourt's pioneering efforts in the authentic-instrument movement from his numerous recordings of works by Bach, Monteverdi, Handel, and Vivaldi. Those familiar with the actual treatises and other historical evidence for performance practice will also know that Harnoncourt, while scoring high in the revival of historical tone colors, often (mis)uses historical evidence to back up weird ideas of phrasing, articulation, and tempo manipulation.

But my reservations on these counts, while not totally allayed, have been thoroughly swamped by the extraordinary musical quality of Hamoncourt's Beethoven, which equals Norrington's as the most interesting cycle since the beginning of the digital era. Unlike Norrington's set (available as separate discs on EMI/Angel), Hamoncourt's Beethoven is not claimed to be in any way "authentic." Modern instruments are used (except for long-bore valveless trumpets), and most tempos are conventional (meaning nowhere near Beethoven's very fast markings). However, Harnoncourt does take all marked repeats and even takes the repeats in the da capo sections of the scherzos (except for the Ninth), granting those movements more prominence than they usually receive.

The orchestral layout is also standard: violins on the left (with the second violins a bit more prominent than usual), cellos and violas on the right, winds in the center, all recorded at "medium" distance in a smallish but reverberant locale (the Stefaniensaal in Graz, Austria). The woodwinds have much more prominence than they are usually granted in Beethoven recordings. While this may be at Hamoncourt's instigation and is at least partially the result of the chamber-orchestra size of the string section, the closer-in sound of the woodwinds compared with that of the strings also indicates the use of slightly overbalanced spotlight mikes. Throughout most of the set, hearing such woodwind detail is beneficial, especially when the wind writing gets complex, as in the second movement of the Seventh and the third movement of the Ninth. But if you didn't know it already, in the finale of the Seventh you'd be hard pressed to hear the looping string figures as the prime motivic material, so prominent are the hiccuping woodwind/brass offbeats.

Although the orchestral layout and recording technique are generally unremarkable, the sound Harnoncourt obtains from this ensemble is definitely out of the ordinary. The "authentic" trumpets he uses have the great virtue being able to cut through without overpowering. Most of the instruments play without pronounced vibrato. This lightens the string texture tremendously, letting the winds peek through. While following along with the scores (Eulenburg

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recording (also on London), the bizarre Harnoncourt rendition, and Klemperer's dialogueless recording. The better "average" performances are those of Levine (RCA), Jordan (Erato), Suitner (Eurodisc), Haitink (EMI), Böhm (Deutsche Grammophon), and Sawallisch (EMI). I would avoid altogether the musical deficiencies of the Koopman recording (Erato—no dialogue and not yet on CD), the heavy-handedness of Colin Davis (Philips), and the over-fastidiousness of Karajan (Deutsche Grammophon). But before you buy that second recording, I would recommend a read-through of the complete libretto. As generations of music lovers have sensed, and as this review has tried to convey, there is much more to *Die Zauberflöte* than meets the ear, even on the best of the recordings.

References.

Score: Mozart. *Die Zauberflöte.* Ed. by Gemot Gruber and Alfred Orel. "Urtext of the New Mozart Edition." Bärenreiter Study Score No.155.

Libretto: Complete Schikaneder libretto with English translation by Susan Webb in *The Metropolitan Book of Mozart Operas.* Harper/Collins, New York, 1991.

Recommended reading (all in paperback).

Branscombe, Peter. *W. A. Mozart: Die Zauberflöte* (Cambridge Opera Handbook). Cambridge University Press, Cambridge, England, 1991.

Chailley, Jacques. *The Magic Flute Unveiled.* Inner Traditions International (a New Age publishing company!), Rochester, VT, 1992. (Previously published as *The Magic Flute, Masonic Opera.* Alfred A. Knopf, 1971.)

Robbins Landon, H. C. *1791: Mozart's Last Year.* Schirmer Books, New York, 1988.

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editions), I could hear no obvious retouchings of Beethoven's orchestration. Harnoncourt has the flutes and oboes take a D in measure 81 of the Ninth's first movement instead of the printed B-flat. Apparently this derives from one of Beethoven's manuscript scores.

As he does to some extent in his recordings of Haydn and Mozart, throughout this set Harnoncourt greatly tones down or smooths the articulation of detached notes (notes with dots printed directly above or below them, indicating that they are to be held for shorter than their nominal value), especially for the strings. These notes have traditionally been played with staccato articulation (sharply, crisply) but Harnoncourt makes them smoothly detached. (It's harder to describe than to hear.) This revision of articulation, together with Harnoncourt's very aggressive treatment of sforzando notes ("forced" or heavily accented notes), creates a very distinctive sound quality to the playing. It is so unique that you'd probably to be able to identify one of these performances from only hearing a few seconds of it.

It is this lean yet powerful texture that gives these performances their special quality, one far removed from traditional (or "authentic") ideals of Beethoven sonority. Time and time again you will hear details in orchestration, phrasing, accentuation, and harmony that are in the score but somehow have never been made evident until now.

Most of the time, the smoothly detached string texture is appropriate, even if it is not what one is expecting (this is a matter of interpretation and not necessarily a violation of the score or of historical authenticity). The smoothed-out articulation does, however, fail miserably during the eight most famous notes of the canon: the opening of the Fifth Symphony. No performance I've ever heard has come close to what I read in the score as two off-balance lurches (Norrington gives it a try). But Harnoncourt's weak opening is definitely underarticulated, almost genteel, in this most violent of movements.

In contrast to the unique sound of the orchestra, Harnoncourt's Beethoven tempos and tempo changes are actually quite traditional. Only a few of his overall tempos are what might be considered unusually fast:

• The opening of the Eroica, which is taken nearly at Beethoven's one-bar-per-second tempo indication (although Harnoncourt slows down, like everybody else, by the end of the exposition).

• The trio in the Scherzo of the Ninth, which is raced through, compared with the tempo of the surrounding scherzo. But this passage, with its exhilarating wind writing, is also one of the high points of the set.

• The slow movement of the Ninth, which may seem fast to some, though Beethoven's markings are faster still. Harnoncourt is good at differentiating the various tempi called for in this movement, which is also a high point.

I found the opening movement of the Pastorale a bit too slow and smoothed out, like a Karajan performance. Here, the smoothed articulation works against the already smoothed-out music. But the remainder of the tempi are very well judged—and traditional.

Tempo manipulations are also in the tradition. Harnoncourt uses rubato or slight tempo changes in all the traditional places, though not as obviously as in some (e.g., the first and second movements of the Eroica, second movement of the Seventh, first movement of the Ninth). Harnoncourt, like virtually everybody else, does not take the double-bass "recitative" in the Ninth's last movement "in tempo" as marked. (This instrumental recitative is *supposed* to sound unnatural in order to give the words *"O Freunde, nicht diese Töne!"* their full force.) At other times he is unusually strict with the score, as in the last moments of the second movement of the Seventh and in the weird ritardando/a tempo cadences in the first movement of the Ninth.

A final point about Harnoncourt's textures: if the picture in the liner notes can be believed, the chorus for the Ninth Symphony numbers only 34 members, this in a piece where whole communities are sometimes marshaled for a performance. The group also sings without much vibrato and absolutely in tune, to produce the some astonishing moments of haunting beauty in the finale of the Ninth *("Seid umschlungen, Millionen!"* etc.). The vocal soloists are generally nondescript, except for the breathless tenor.

In all, Harnoncourt's Beethoven is a masterful achievement, combining almost conventional tempos with a thoroughly recast orchestral sound. It is a "must hear" to all who think they know how these pieces should go. If you don't want them all, hope for separate releases of Symphonies No. 3 and 9. They are the standout performances in an already noteworthy cycle. •

Editor's choice of recent, sonically outstanding CDs (to be fully reviewed next time):

Bach, J. S.: "Switched-On Bach 2000" (All-New 25th Anniversary). Wendy Carlos. **Telarc** CD-80323.

Bartók, Béla: Music for Strings, Percussion & Celesta; Concerto for Orchestra. Orchestre de la Suisse Romande, Eliahu Inbal, conductor. **Denon** 81757 9044 2.

Brian, Havergal: Symphony No. 1 ("Gothic"). CSR Symphony (Bratislava), Slovak Philharmonic, soloists, choruses, Ondrej Lenard, conductor. **Marco Polo** 8.223280-281.

Offenbach, Jacques: Gaité" parisienne (complete); other works. Ibert, Jacques: Divertissement. Cincinnati Pops Orchestra, Erich Kunzel, conductor. **Telarc** CD-80294.

Piston, Walter: Symphony No. 4; Three New England Sketches; other works. Seattle Symphony, New York Chamber Symphony, Gerard Schwarz, conductor. **Delos** DE 3106.

Schuman, William: "A Tribute to William Schuman" (four different works of the composer). Seattle Symphony, Gerard Schwarz, conductor. **Delos** DE 3115.

Shostakovich, Dmitri: Symphony No. 7 in C Major, Op. 60 ("Leningrad"). Dallas Symphony Orchestra, Eduardo Mata, conductor. **Dorian** DOR-90161. *[A1 sound!—Ed.]*

Vaughan Williams, Ralph: Symphony No. 6 in E Minor; The Lark Ascending; Fantasia on a Theme by Thomas Tallis. BBC Symphony Orchestra, Andrew Davis, conductor. **Teldec** 9031-73127-2. *[Great music, too!—Ed.]*

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The Audio Critic

In the next issue:

We return to the audio component most in need of reviewing, the loudspeaker, with a spate of test reports on units priced from the middle three to upper four figures.

The promised guest article on very high-efficiency speaker systems, dropped from this issue because of its length, makes its highly stimulating appearance.

We return to another subject we have neglected for a while, power amplifiers (yes, with David Rich on deck).

The tail end of the preamplifier survey is squeezed in (meaning just a few more units plus mop-up comments).

More attention to home theater systems, audio/video, etc., including the promised review crowded out of this issue.