Issue No. 20

The Audio Critic®

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We launch our power amplifier survey with seven reviews for openers, circuit critiques, a new method of measuring dynamic performance into complex loads, and an amplifier tutorial by Dr. Rich.

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We report our initial tests of the DCC system.

Part III of the Ranada interviews with the elite of audio brings you yet another original thinker.

Plus other test reports, editorials, our famous "Hip Boots" column, letters to the Editor, and lots of CD reviews.

Issue No. 20 Late Summer 1993

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For subscription information and rates, see inside back cover.

Contents

From the Editor/Publisher:
Are We at All Catching Up with the Schedule?

This is always the last page I finish before a new issue goes off to the printer, and as I am writing this we are well into September but technically it's still summer. (A couple of days ago we had one of the hottest days of the year here in Bucks County.) Now, this is the Summer 1993 issue—for safety I labeled it Late Summer 1993—and I am hoping that, technically, it doesn't become an early fall issue. That would create a traffic jam because the real Fall 1993 issue will also have to be published before the fall is officially over; at this point late November is the absolute best we can hope for, and it will more likely be early December (I just hope it won't be snowing yet). Obviously this is not the most desirable state of affairs, but somehow our quarterly must keep up with the four seasons of the year, and if it does we will have made some progress.

The latest development is that I have found a highly competent, widely respected, literate, and definitely "nontweako" audio journalist whose name will go on the masthead of the next issue as Assistant Editor. This savvy professional may actually be able to do half of my work, in which case it should take only half as long as before to finish each issue. 'Tis a consummation devoutly to be wish'd. Lest I should jinx a situation with a previous history of bad luck, I am withholding the name until Issue No. 21 is a reality. (Who says I can't be superstitious just because I'm not a tweako audio cultist?)

While on the nagging subject of our publishing schedule, I want to respond to a very small number of nasty letters whose tone renders them unfit for publication but whose message should be addressed. How dare we use a slogan like "Accountability in audio journalism," these fulminators write, when we are late all the time? Well, to me accountability in audio journalism means that we carefully measure the audio components we review in a well-equipped laboratory, do our comparative listening double-blind at levels matched within ±0.1 dB, and have our technical discussions double-checked by graduate engineers. Other "alternative" audio publications fail to do that. Accountability in publishing, on the other hand, means that we deliver four big, fat, lovingly edited issues for every $24 you send us, whatever the interval between them, and cheerfully refund the unused portion of your subscription if you ask for it. When it comes to speed of publishing, I must respond with the words of the bumper sticker I saw recently on an old car: "Bear with me—I'm pedaling as fast as I can."

Arrivederci in... well, late fall.

Box 978

Letters to the Editor

This is a column for letters of general editorial interest. That classification covers a lot of ground but emphatically does not include things like "I have a Schlockmeister M-100 power amplifier and I'm wondering if I should move up to something better—what do you folks recommend?" Not that we answer letters like that privately, either, because our business is magazine publishing, not private consulting. 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:

At the last two Consumer Electronics Shows (Summer 1992 and Winter 1993), room acoustics correction devices based on digital signal processing (DSP) were shown by Snell Acoustics. The first time Snell used DSP from SigTech and the second time they had their own, developed with Audio Alchemy (the tweako digital jitterphobes). The second time, SigTech also displayed independently. The demonstrations were accompanied by pushy proclamations of a "revolution in sound." One repeating claim was that the processors "make the room disappear." When the hype stopped and I was able to listen to their contrived A/B comparisons, I was not impressed.

I would like to use your magazine as a forum to state my concerns about this technology and to, hopefully, get some questions answered.

DSP questions:

Do responsible people on the inside, like Kevin Voecks of Snell, really believe all the hype? Can DSP fix a room with poor acoustics to make it just as good as one starting with good acoustics? Can DSP create an acoustic environment better than any possible real room? If the DSPers will commit to a "yes" answer to any of the above, then when can we test a conventionally optimized system/room against an "improvement" by adding DSP? Is there a reason why this preference test cannot be double-blind?

DSP concerns:

DSP room correction is different from conventional room equalization in that it is able to make corrections in the time domain as well as the frequency domain. Actually, fully correcting the amplitude versus time (time domain) measurement of a signal automatically corrects the amplitude versus frequency (frequency domain) measurement, but the converse is not true. Conventional equalizers do a good job of correcting problems produced by the speaker system but not problems produced by room reflections.

Consider a speaker, a listener, and a single reflecting wall. Sound from the speaker travels both a direct path and a (longer) reflection path to the listener. The acoustic pressures add at the listener, but not in phase because of the time difference caused by the path length difference. The resulting frequency response is a series of peaks and nulls called a "comb filter" response. For steady sine waves, this could be equalized to flat, but that would not fix the basic time domain problem, which is two arrivals.

Now let's deal with the problem in the time domain, using DSP. The problem is now simply the two arrivals, not a complicated comb filter response. Since DSP makes us time-domain-agile, we will feed our speaker a delayed and polarity-inverted version of the original signal. This will arrive at the listener just in time to acoustically cancel the signal reflected from the wall. Now the sound is "equalized" in both the time and frequency domains.

Perhaps you already see the ugly gremlins lurking in the background of this happy picture. First, the delayed signal designed to cancel the reflected sound also creates its own reflected sound, which needs to be canceled. This could go on forever except for the finite number of instruction steps available in the DSP. Fortunately, because of the longer path length of the reflected sound and because of imperfect reflection, the reflected sound is not as loud as the original. This means that each correction signal is of a smaller amplitude than its predecessor. Hopefully, by the end of the processing time (900 milliseconds for the Snell system), the umpteenth correction of the correction is small enough to be neglected.

Let's try an example. Say the speaker is ten feet away and a reflection from a plaster wall takes an additional two milliseconds to arrive. By my calculations, the reflected-sound SPL would be 1.86 dB lower than the direct. Thus, the inverted canceling signal should also be 1.86 dB less than the original. Its own reflection would be down another 1.86 dB and so on. After 54 corrections of the corrections, the amplitude would be more than 100 dB below the original—small enough for further corrections to be neglected. At 2 milliseconds per correction, this would take only 104 ms, well within Snell's processing time.

Of course, the real listening rooms we want to correct are much more complicated. Acoustic energy radiated by the speaker does not simply encounter one or more early reflections, pass our listener, and disappear. The walls contain the acoustic energy within the room, allowing it to leak out relatively slowly. This is measured by reverberation time.

Let's consider a very simple sound from one speaker, a tick approximating a mathematical impulse. This tick of energy radiates in all directions covered by the speaker, encounters walls, and continues on (attenuated by the reflection) to encounter more walls. For the listener, the density of the ticks arriving increases with time, while the average amplitude of the ticks is reduced with time. The increase in density and decrease in amplitude are a function of room characteristics, particularly diffusion. For the usual listening room, the reflection density increases geometrically to extremely high values. A DSP acoustics corrector must be programmed with this "impulse response" in order to cancel it and to cancel a portion of the infinite series of equally complicated corrections of the correction signals.

Not only is it an awesome task to correct for a simple "tick" in a room, but such correction must be performed continuously on two channels of music in real time. We can label ourselves gullible if we believe this without asking for proof that it has actually been accomplished.

Next we have the matter of what is meant by "acoustic cancellation." Here, it means that at some point (where the listener is), the pressure components of two sound waves cancel. Note that this is only at a point (or possibly a line or plane). The total acoustic energy in the room is unaffected by this local cancellation. Consider that we are dumping considerable energy into the room to achieve cancellation at a point. Even if cancellation at the listener is perfect for 900 milliseconds, won't this reverberating sound byte garbage be audible after this time?

Even in the aforementioned simplified situation of a speaker, a listener, and a wall, the total energy of the 104 milliseconds of corrections for the corrections is more than 1.34 times the original sound energy. This reverberant sound is not canceled; it is a problem which is simply pushed back in time. When the time is up we have a bigger problem.

Imagine the tick with one reflection. For 900 milliseconds after the original sound, the speaker emits a decaying series of in- and out-of-polarity ticks totaling more than 1.34 times the energy of the original, for the purpose of preventing the listener from hearing a single reflection. We must demand a demonstration that this is possible. We must demand a demonstration that the dumping of additional energy into the room has made the acoustics better, not worse.

Now we come to another point (literally). This is the "point" at which the cancellation occurs, the location of the listener. Listeners have two ears separated by six or eight inches, depending on whether you measure through or around the head. Acoustically, these cannot be considered to be the same point except at the lowest audible frequencies.

Example: Two pressures are equal and opposite at the left ear of a listener, say, arriving from one of the two stereo speakers and a sidewall reflection. The arrival-time difference for the two ears is likely to be around 0.4 millisecond. Therefore, at the right ear the phase of the problem signal and its canceler will be shifted, resulting in imperfect cancellation. At 200 Hz, the cancellation at the right ear will be only -18 dB, not a figure that is really acceptable like -100 dB. Remember, this is at 200 Hz. Lower frequencies will have better cancellation, but the effect will deteriorate at higher frequencies until, at 600 or 800 Hz, there will be no cancellation at all.

The example was for a single listener optimally positioned in the sweet spot. For listeners who move or rotate their heads, the cancellation will be diminished. For other seating positions, all bets are off.

Stepping back from implementation issues for a moment, we might ask if the room-correction goal of acoustically "removing the room" is a worthwhile one. Real sound in a concert hall can be thought of as having infinite channels—sound comes from every direction. We use two channels for practical reasons. The fact that we have two ears is no more an excuse for limiting ourselves to two reproduction channels than two eyes would be an excuse for using two spotlights to illuminate a room. Two channels are an okay compromise only because we have a listening room to supply delayed sound from other directions. Remove the listening room by using an anechoic chamber and you have bad sound. So, let me ask the question: Assuming DSP room correction can be successful, do we *want* our listening-room reflections removed?

Snell's earlier demonstration offered an A/B switch between straight through and processed. I thought I heard improved "equalization" via processed, but accompanied by a distinct "computer sound" echo on male speaking voice. This echo was not audible to me on the music that I tried. The second showing did not offer an A/B comparison. The reason why not was said to be too complicated to explain. I remain a skeptic and I demand A/B comparisons to be convinced there is merit to this new DSP application.

Sincerely,
David Clark
DLC Design
Farmington Hills, MI

/ was eagerly looking forward to the Snell demonstration at the Summer CES (June 3-6, Chicago), but then Snell canceled their participation in the last minute—one of the many letdowns of the show—and none of us got to see and hear the latest version of their DSP technology. The reasoning behind your reservations is powerful and convincing, but let us wait until everybody's cards are on the table before we pass final judgment.

—Ed.

The Audio Critic:

It is with great trepidation that I enter this fray. Especially after making a significant sacrifice to put a Krell amplifier into my system because it did musical things for me that no "lesser" product could do. Moreover, I can describe the essential superiority in unambiguous musical terms.

A bass soloist, singing a Handel aria, was having professional difficulty with a low note. Through comparable amplifiers, this fact was obvious with Krell, detectable with PS Audio, and inaudible with Adcom. This hierarchy was clear, unmistakable, and unaffected by relative volume levels. This one note ultimately led to my purchase of a Krell amplifier.

In an effort to rationalize my action, I have come to refer to this characteristic as "resolving power"—the ability of an amplifier to respond to and reproduce the most subtle forms of musical information. It was very clear with the three instruments involved that the threshold of "resolving power" was different in each case.

It follows that if a specific musical event is masked, so will all the musical information below the threshold of resolving power represented by that event. I am convinced that such information is what is often referred to as air, depth, spaciousness, ambience, focus, soundstage, and the like. And, indeed, my general subjective reaction to the three instruments mentioned would have to be described using such terms. So, too, would the differences between my Krell amplifier and the one it replaced.

What's the point? Unless semantics are playing tricks with me, my observations differ from many I have come across over the years, and they might just be of value.

1. "Definition" and "resolving power" are not the same. Adcom, PS Audio, and Krell are fine, high-definition amplifiers with varying degrees of resolving power.

2. Competently designed and manufactured modern amplifiers do not sound the same. In the absence of an appropriate and definitive musical event, however, the differences are difficult to identify reliably and describe objectively.

3. If a difference can be heard, it can be measured, but one needs to know where to look. Frequency bandwidth, power bandpass, and freedom from noise and distortion all help, but mostly with different and less subtle problems. Information relating to resolving power, on the other hand, is audible even at the lowest listening levels and represents a tiny portion of an already tiny signal.

4. If I were looking to define resolving power, objectively, I would do two things: (a) Perform a circuit design and execution analysis of comparable Adcom, PS Audio, and Krell instruments to determine what causes the progressive masking of extremely subtle musical information. (b) Look to measure differences in the behavior of the instruments as the informational content approaches zero—both with respect to threshold and speed of response.

5. Let me concede that I also admire Krell products for the beauty of their industrial design and build quality, and accept the effect of these factors upon their cost. How nice it is that both physical and musical prowess can coexist in the same instrument.

I am directing this admittedly self-therapeutic letter to you because my reading of your journal suggests that you, particularly, may respond with interest.

Sincerely,
Paul P. Siegert
Geneva, IL

You're a dangerous man, Paul, a potential corrupter of the minds of novice audiophiles who might cross your path. You're obviously intelligent, articulate, analytical, self-confident, familiar with the audiophile scene, and therefore plausible—but you happen to be 100% wrong! What you write is basically tweako rubbish, put forth with a bit more class and suavity than I generally see in my day-to-day tweako mail. There is no such parameter as your "resolving power," distinct and separate from frequency response, noise, static and dynamic distortion, power-supply characteristics, etc., etc. Your suggested investigations under 4(a) and (b) are inherent in our current test protocols. There is nothing in the electronics textbooks, nor in the IEEE or AES literature, to support your tiny-portion-of-an-already-tiny-signal nonsense.

The Krell vs. PS Audio vs. Adcom experience you describe took place, I suspect, in an audio salon—didn't it?—under the tutelage of a "coach" (i.e., a salesman who wanted to sell you a Krell). If you had reported that you were able to pick out the Krell double-blind, at levels matched within ±0.1 dB, seven times out of eight replays of that low note of the bass soloist, then I would begin to believe that your perceptions had some validity. Since you say that the differences were "unaffected by relative volume levels," I know right away that there was a lot of completely undisciplined volume-control twiddling, and it's also quite apparent that the test wasn't blind. You went through a meaningless exercise, Paul, and ended up with an extremely expensive amplifier that in reality sounds no different on that low note but is well-engineered, beautifully built, and solidly backed by a reliable company. Worse things could happen to an audiophile, so you really don't have to rationalize your compulsive purchase with science fiction.

One more observation. When I hear the word "speed" applied to the reproduction of instruments by an amplifier, I know I'm in tweako territory and start walking rapidly toward the border.

—Ed.

The Audio Critic:

It's always fun for me to read your witty and often acrid writing—except when I think that your remarks might be aimed at me. I'm referring to the admonishment to your readers, inserted just before the "Letters to the Editor" section of Issue No. 19, that many of us have written "letters that should never have been sent..." Since I recently wrote you a letter that I consider to be "intelligent, well-informed, and well-written" (your words), but which was not published, I cannot help but wonder if I am among those committing "reciprocal punditry."

No, my feelings aren't hurt because my letter was not published, nor will they be if this one is not. The topics and questions in my letters may not be interesting to other readers, in your opinion, and that's OK, because it's your magazine. But I do think that this letter should be sent, because you are treading on thin ice when you seem to insult a broad, unidentified spectrum of your readership, and you also run the risk of causing the timid among us never to write a letter. Your magazine cannot survive without subscribers.

Certainly you may, and should, expose fuzzy thinkers and deflate pompous "experts" in your inimitable style, because that is why, I think, that many of us subscribe to *The Audio Critic.* But I hope you can confine your attacks to specific letters, articles and advertising claims, and not sweep all us unpublished letter-writers aside because we are cluttering your mail...

Sincerely,
Sheldan C. Collins
Weehawken, NJ

It never ceases to amaze me how touchy some people can get when they perceive, reading a totally impersonal journalistic generalization, that somehow "the shoe fits." In your case it fits twice, both in the above letter and in the one that wasn't published. The above letter extensively quotes my introduction to this column in the last issue but characteristically omits the crucial sentence: "Because they ask questions and bring up arguments that have already been answered in our pages... " I.e., not because you're "cluttering" our mail—wasn't that quite clear and readable? So once again it appears that you'd rather write than read. Your unpublished letter asked questions about power line conditioners to which you could have read the answers, perhaps not in minute detail but certainly in all essentials, on page 60 of Issue No. 16. Your subscription record indicates that you have that issue.

Anyway, you have succeeded getting yourself published, although you're probably not satisfied with the context. Thank you, in any event, for your complimentary remarks—and don't feel inhibited about writing, as long as it follows attentive reading.

—Ed.

The Audio Critic:

Here's what I call "News from the Moronosphere."

Speed of light c in vacuum:

2.997924563×10^8 meters per second, or 186,282.396 miles per second.

Propagation speed of electrical signal through wire (according to Dr. David Goodstein, Chancellor of Cal Tech and Professor of Applied Physics): between 0.8c and 0.6c for theoretically "pure copper" and junk wire, respectively.

Using the slower speed to obtain the shortest wavelength from a given signal:

186,282.4x0.6=111,769.44 miles per second.

For wavelength, we divide propagation speed by frequency. If we use 20 kHz as the highest frequency at which audiophiles can "hear" phase shift, we get:

111,769.44÷20,000=5.59 miles, or 29,507 feet (per each 50 µs).

A quarter wavelength at 20 kHz is 1.4 miles long in the wire.

And finally, dividing the wavelength by 360°, or one entire cycle, we get:

29,507÷360=81.96 feet.

A speaker cable long enough to exhibit 1 degree of phase shift at 20 kHz would be 82 feet long!

Even in the face of incontrovertible proof such as this, these boneheads still insist that they can hear phase anomalies in 30 feet of cable. It makes you wonder if they even know what "phase" is.

Drew Daniels
North Hills, CA

What they surely don't know is what "hearing" is. They think hearing means telling yourself and your friends that you're hearing something. You don't have to prove it; you just have to assert it. And you can't be tested to find out whether you're really hearing what you claim to be hearing because the test will inhibit your hearing ability. It's a closed system and it seldom gets to the point of scientific analysis, such as you present.

—Ed.

The Audio Critic:

...audio professionals think I'm crazy because I like to play with tubes! Can't help it, can I, if I happen to *like* a little tube distortion? I know you think tube audio is nonsense, but hear me out.

On page 5 of your Issue No. 19, you say that tube audio "has little or no support in the professional engineering community." Please see the enclosed articles from *Mix* and *R·E·P*. The use of tube audio processing is a major part of the professional recording business; highly respected recording engineers and producers go shopping for a studio partly on the basis of its supply of tube compressors, equalizers, microphones, and sometimes even mixers and tape recorders. Few or none of these pros are tweak audiophiles; in fact, most of the ones I've met over the years have never *heard* of "high-end" audio.

Not all tube companies are "tweaky" —outfits like AKG, Sennheiser, Summit Audio, Tube-Tech, Demeter, EAR, and some others are making tube electronics *strictly* for the pro audio world These people use tubes for their peculiar distortion behaviors and their dynamic-range capabilities. None of this has anything to do with the tweak world. (Although EAR does make the usual overpriced amps and preamps for high-end sales, they also make more conventional studio equipment, such as mic preamps, using tubes.) Not to mention VTL's Manley line.

And don't forget the tube guitar-amplification market, more than $100 million worldwide last year and growing after several years of decline. Guitarists tend to like that tube distortion, plus have a legitimate need for amps that can take overloads without damage and without the interference of protection circuits. (Rock guitarists have a tendency to destroy amps that go into protection in the middle of a song.)...

The best part of making tubes a hobby is the continued availability of good, low-cost tubes. I don't really agree with you about them wearing out, at least preamp tubes. Good ones will give full gain for 100,000 hours and more, routinely. That's almost 12 years of continuous operation—I routinely see old hi-fi equipment, with preamp or driver tubes 30 and 40 years old that still work like fresh-from-the-box tubes. Power tubes are a bigger problem; typical beam-power or pentode power tubes are only good for 4000-5000 hours at best. This is why, if someone who comes to me wants to explore the tube sound, I tell them to start with preamps—tube power amps require too much maintenance for most people. But when the tubes do wear out, there are plenty of reasonably priced replacements coming in from China, Russia, and the Czech Republic, and the supply is expected to continue for a long time—mainly to fill the needs of the guitar amp, pro audio, and industrial markets....

Who am I? I do have a B.S.E.E., so snow jobs carry no weight with me. I'm a contributing editor at *Glass Audio,* probably the only regular contributor who doesn't tweak or believe in tweaking. (Most of *GA's* construction projects are amps and preamps that are made for extreme accuracy; i.e., they end up sounding like solid state.) That's a whole strange political situation; I regularly get attacked for saying things like "why bother making a tube circuit sound perfect, unless it's just a technical exercise? If you want perfect, transistors are more practical. If you use tubes, use them for their own unique characteristics." Some of Ed Dell's subscribers are themselves would-be tweak gurus, and my position is upsetting to them...

Good luck with your magazine. If

those jerks in Santa Fe can reach 70,000 readers, then you can too. More easily I think.

Sincerely,
Eric Barbour
Albuquerque, NM

What you say is absolutely true, but what I said was also true. I should perhaps have written "professional E.E. community" instead of "professional engineering community," thereby excluding the recording engineers, who can be every bit as unscientific as the tweakiest audiophiles. They get paid for their ears, musical taste, and hands-on experience, not for their profound insight into electronic circuitry.

When a recording engineer swears by a piece of classic tube gear, it may very well be that no exact solid-state equivalent of it is available and that the manufacturer keeps selling it year after year; it doesn't follow, however, that no exact solid-state equivalent of it can be designed by a competent E.E. If you ask an E.E. professor at the Massachusetts Institute of Technology to design for you, say, a compressor—or even a guitar amplifier—from the ground up, he isn't going to reach for the tube manual. He will do it with solid-state devices, and he will be able to give you any "peculiar distortion behavior" and any "dynamic-range capability" you want. That was all I was trying to say. I never said there was anything inherently wrong with tube equipment already in use in someone's studio or home.

Yes, an old 12AX7 is much more likely to be in working condition than an old 6L6, but a 741 op-amp will easily outlive them both, barring some sort of catastrophe.

As for the Atkinson, Archibald, and Santa Fe, my tracks are laid elsewhere and go to different destinations, so it's hard to make projections about future passenger traffic. My little engine keeps saying, "I think I can!"

—Ed.

The Audio Critic:

Your magazine is a true pleasure to read. Dr. Rich is a wonderful addition.

Sincerely,
Hillel J. Kumin
Associate Dean
College of Engineering
The University of Oklahoma
Norman, OK

Confucius say, "One gracious compliment from engineering-school dean worth suffering one thousand indignities from untutored tweaks." Thank you.

—Ed.

The Audio Critic:

I was delighted to see you *[namely the Ed.]* again in Chicago; I trust you had a fine time carrying the fight against audio fallacy into the hotel suites of the ignorant. ...

I am certainly a fan of *The Audio Critic;* in addition to its fine reviews and tutorials, I admire its stand against the uninformed and feel privileged whenever my name is invoked against the forces of sloth and stupidity. Please keep up the good work. The audio industry and its many enthusiasts need your help.

Sincerely,
Ken Pohlmann
Professor of Music Engineering
University of Miami
Coral Gables, FL

As an audio journalist and editor I try to earn the respect of academics and professionals who know more than I do, such as you, Ken; the semieducated gurus of the various branches of high-end audio obviously seek the respect of those who know less than they do. I think that's a defining distinction, and it's great when someone with your credentials confirms it—so I'm the one who feels privileged.

But what's this sloth business, Ken? Every tweako audio journalist I can think of is busy as a bee; it would be nice if they were a bit slothful now and then.

—Ed.

The following six letters comment, pro and con, on the general subject one could call (with somewhat superficial brevity) "tweak bashing" or "ignoramus hunting." The Editor responds to them as a group.

The Audio Critic:

.. .In these post-Enlightenment times it is alarming to note the general spread of know-nothingism in many areas of life, and particularly to see voodoo practitioners attempting to pass themselves off as scientists and engineers. *The Audio Critic* provides a welcome rational voice in the audio community....

...I do wish that you would cease and desist with the "Hip Boots" column. It's not that I think it's not both informative and amusing, but it is rather like watching a microbiologist explain bacteria to a witch doctor. I personally would prefer to see more time, energy, and space devoted to information than to rebuttal of misinformation; besides, I believe Darwin was right, and with any luck the charlatans and snake-oil *salespersons* (just to be fair to EL) won't reproduce....

Sincerely,
Terrence McCarthy
Brooklyn, NY

The Audio Critic:

...Your "Hip Boots" columns are superb. Is there any industry dominated by bullshit as much as high-end audio? I subscribed to *The Absolute Sound* almost from the first issue, until I realized that Enid Lumley was not a self-parodying invention of the reviewers. Then I realized that a company like Adcom could market the perfect amplifier (they haven't) at $800 list and not do better than Class C in *Stereophile.*

You would do your readers a great service if, as sort of an adjunct to "Hip Boots," you would list all of the gadgets called "accessories" in the high-end press and indicate whether the products are very useful, not worth the money, or a downright fraud. You wouldn't even have to mention brand names, since we all will identify the products.

Keep up the "good worth." Your publication is a real bargain.

Edward Doyle
Weaverville, NC

The Audio Critic:

...I have read your most interesting Issues No. 16, 17, and 18, and am impressed with the objectivity with which the products are evaluated. However, temperate use of language may be far more effective than the scathing attack on those who do not review the products as objectively as you do....

C.K. Vissanji
Bombay, India

The Audio Critic:

Keep 'em coming. After Issue No. 10, I canceled my subscriptions to *Stereophile* and *Hi-Fi News & Record Review.* The Audio Critic is the "smart bomb" of audio journalism—always right on target and always devastating.

Sincerely,
Chris L. Walker
King of Prussia, PA

The Audio Critic:

I have decided not to renew my subscription to *The Audio Critic.* At first, I rather enjoyed your editorial style. Four issues later, I think the constant put-downs of the two other audio journals, their editors, and writers have become very tiresome.

Very truly yours,
John Overman, M.D.
Independence, MO

The Audio Critic:

As a new subscriber to your magazine, I recently received, and read, Issues No. 16, 17, and 18....

The way you test audio products, with the emphasis on the facts and the use of a reliable, scientific, reproducible method, is very professional, useful to your readers, and refreshing.

Before going further, I feel that I must tell you a little more about my background and my experience in the audio field.

In 1978, I created a company, called *Architecture & Physique Appliquée.* Our products were sold under the name Goldmund, until the beginning of 1981. The company (i.e., the name Goldmund, and the right to produce and sell the products) was then sold to our former distributor.

Although we were a small company, we had the opportunity to create and develop new concepts, new techniques in high-end audio, thanks to the use of specialized consultants, with computing facilities.

We first introduced the Goldmund T-3 radial tone arm. This unit was opto-electronically controlled, so as to ensure an almost perfect position of the cartridge during the record play. A turntable was introduced the following year, called the Goldmund Studio, combining for the first time the direct-drive principle with the floating subchassis technique. In addition, the use of a high-inertia platter, machined in methacrylate and lead, allowed a very low level of resonances. The last *Architecture & Physique Appliquée* product was the Classique preamplifier, with very high slew rate, 75-volt power supply, and very short signal path.

After the company's closing, in 1981 (a short life, due to our lack of commercial talents), different products continued to be marketed under the name Goldmund that were not designed by my team any longer (but the brand *Architecture & Physique Appliquée* remained my property and was thus removed),

We were happy to find that some well-known "audio critics" rated our arm and turntable at the "state-of-the-art" level (in *The Absolute Sound).* I still don't know what were the comments and evaluations of our products in the rest of the audio press in the U.S.A. (not being easily available in France at the time)....

Although the tests in *TAS* didn't seem to be conducted in a very scientific way, we were, of course, glad to see that our methodical research and calculations were confirmed by an independent reviewer (that is, independent of the company at least), even subjectively...

I would like to add a suggestion to this letter.

I think all your readers are satisfied with your well-informed, accurate, and reliable articles. But some, as I do, may think that too much space is dedicated to contradict and debate with the "subjectivist" brigade. You can assume that the vast majority of your readers are aware of what a fair, reliable test or comparison is and, more important, are not that much interested in knowing exactly when and where Mr. X, or Y, was wrong in his hype and unscientific evaluations.

What a good, competent reviewer can do is to use his knowledge, laboratory, and experience with audio products to go straight to the point, and extract the valuable information out of the facts. This is, in my opinion, what your readers find most instructive in *The Audio Critic.* Why should you waste time arguing with astrologers and fortune-tellers (so to speak)?

I hope these few lines will be of some use to you and your team... Congratulations on your work.

Yours faithfully,
Michel Levy
Paris, France

So—how should we deal with the witch doctors, charlatans, snake-oil peddlers, Enid Lumleys, frauds, "other" journals, astrologers, and fortune-tellers of the audio world (to borrow our correspondents' terminology)? Should we devastate them, in the words of Chris Walker and with the concurrence of Edward Doyle and even Prof. Pohlmann (further above)? Should we avoid all confrontation with them because it's unseemly, as Dr. Overman and C.K. Vissanji appear to believe? Or should we just give up on them and attend to more important matters, letting them slowly but inevitably self-destruct, as Terry McCarthy and Michel Levy seem to think is wisest?

I think the underlying question here is just how influential and effective this untutored, antiscientific element has become in audio, and I think the answer is: very. Go to your local audio salon and find out where the pimply-faced "audio consultant" in the Metallica T-shirt gets his strong opinions. From Professors Stanley Lipshitz and Richard Greiner? No way! From Harry Pearson, Robert Harley, and other loudmouthed tweako "authorities." Those strong opinions are then imparted to that nice retired dentist with $20,000 to spend on a new system, who probably has never heard of Lipshitz and Greiner—or, if he has, thinks they represent just another partisan opinion—and who will then impart them to all his well-heeled friends. He is the key player in this situation.

You're wrong, Messrs. McCarthy, Vissanji, and Levy; ten years and more could go by before that dentist and his friends realize—if they ever do—just how stupid and ridiculous those tweako opinions are, unless I and my colleagues are constantly in their face with the documented, tweak-humiliating truth. And even then...

My decision is to heed the advice of Ken Pohlmann. In the immortal words of William Blake,

I will not cease from mental fight,
Nor shall my sword sleep in my hand

until I see the entire tweako cultist scene discredited in the eyes of the majority *of audiophiles, not just the professionals and academics.*

—Ed.

The Audio Critic:

As a new subscriber, I am somewhat disappointed in your reviews. Before I mention specifics, let me state that I am not an "expert, tweak, geek," or whatever term one chooses to use. I do have minimal knowledge about electronics and speaker systems.

I would appreciate your response to the following, which concerns your Hsu Research HRSW10 review in Issue No. 19:

1. You state, "The enclosure is... acoustically inert...." Olsher *(Stereophile,* March 1993) says, "...the SW10 was alive to the touch...a nut driver I'd left on top of the enclosure started to dance in rhythm to the signal. It felt as if

the whole enclosure was readying itself for launch."

You say, "The distortion may have been slightly higher than with the Velodyne ULD-15 Series II at equal SPLs...." Keele *(Audio,* November 1992) has THD graphs which show a *maximum* distortion of 2% in the 20 Hz band of the Velodyne, but the Hsu graph shows much higher levels (up to 15%).

The above concern is related to subwoofers for one reason: I want to buy a good one; it's not that I'm picking on Hsu (or anyone, for that matter).

I do appreciate your straight talk about the "pseudohigh-enders" who claim to hear what most others cannot.

I look forward to your response.

Sincerely,
James R. Story
Miami, FL

I am genuinely distressed about your initial disappointment as a new subscriber, even though your negative reaction represents about one in a thousand. It may be, however, that you don't read our reviews attentively enough—nor the reviews in Stereophile *and* Audio.

I didn't write that "the enclosure is...acoustically inert...." Ellipsis has its limits. I wrote that "the enclosure is a cheap but extremely strong and acoustically inert paper tube...." The paper tube is indeed acoustically inert. Olsher himself (not that I have the slightest respect for him as an audio reviewer) wrote, "The weak spots are the end caps... .A tube readily resists radial pressure...," etc. So we have no momentous disagreements about the tube. About the end caps Olsher is wrong, as usual. The end caps are quite small, stiff, and essentially mode-free; they are in a sense the "braces" of the structure. The problem is that the entire structure is very light, so that high-energy axial excitation (i.e., long-excursion woofing) makes it "dance" a bit—as a unit. This is analogous to the recoil of a lightweight weapon using heavy ammunition. If the enclosure were bolted to a concrete floor, the effect would be minimized. Even Olsher (there I go again) refers to this "grounding" consideration. The sonic consequences are negligible, in any event.

As for Keele (for whom I have the highest respect), look again at his words and pictures. First of all, he compares the Hsu HRSW10 with the Velodyne F-1500, not the ULD-15 Series 11. Secondly, he points out that "the maximum levels [of the F-1500] were set by the action of the system's limiter circuits." For example, at 16 Hz and 20 Hz, the F-1500 cuts off at 88 dB SPL and 2% distortion. At that level and those frequencies, the Hsu is barely higher in THD. What I wrote does not contradict Keele, even if you assume that the ULD-15 Series II and the F-1500 have absolutely identical distortion characteristics, which is not the case.

My recommendation: If price is no object, or of relatively small importance, get a Velodyne—because it's closer to perfection. If price does matter, however, then get a Hsu (get two Hsus, that is) and you'll be just as happy.

—Ed.

The Audio Critic:

It was with some difficulty that I waded through the rarified atmosphere of "...the longest crank letter ever..." in Issue No. 18. As I struggled, it occurred to me that, for a follow-up, what is needed is the shortest crank letter ever.

Much like Ye Ed., I thought I detected a slight difference on some recordings when inverting polarity. Rather than being confused by a bunch of papers on psychoacoustics, I elected to conduct my own experiment, starting a the point where sound terminates and then working backwards from there.

1. A good flat mike was placed at listening position (only place that counts).

2. Room had already been neutralized with my version of a poor man's anechoic chamber, so received sound was a virtual sonic copy of speaker output.

3. Speaker geometry was optimized for best soundstage and flattest response.

4. A test signal was applied through a switchable inversion stage to line amp and was found, at speaker input, to be a precise mirror image (equal shape and size) when inverted. Positive-going pulse was identified as a speaker push and negative as a pull.

5. Test signals which appeared at mike were *not* precise mirror images when inverted but indicated more compression in the negative direction than the positive, the effect becoming more noticeable as drive was increased.

6. Newly built and hopefully more linear speakers were deployed, and inversion differences essentially disappeared—distortion increased equally as overload was approached.

In the 8 years I've lived with this system, only one person has heard an inversion difference. His auditory senses obviously were quite acute, as he heard a difference when I only pretended to invert.

Here on the low plains—as opposed to the hill country and University of Texas—we refer to it as the "cow chip effect" and let the chips fall where they may.

Donald F. Scott
Houston, TX

1. Ye Ed. used to be Gordon Holt's way of referring to himself, if I remember correctly; I try eschew such cutesyisms. (I'm not nearly as cute, cuddly, and pixieish in my autumn years as Gordon.)

2. Cow chip effect vs. B.S.—isn't that sort of analogous to the politically (not theologically) supercorrect way of referring to the Deity as She? We must not be sexist, or gender-insensitive, even when it comes to bovine excreta.

3. What you say about polarity inversion is quite consistent with Prof Richard Greiner's highly researched paper on the subject at the 91st AES Convention (October 1991) in New York.

—Ed.

The Audio Critic:

I am...a loudspeaker hobbyist. That means I design and build my own systems and basically have a hell of a good time tinkering around with them. I agree that the loudspeaker is the most important part of the audio chain....

...I received some literature from Martin-Logan the other day. Nice-looking speakers. But, I found something in their specifications that tickled me and that I thought you might appreciate (see enclosed).

[The Martin-Logan literature in question claims the following specification for the 12" woofer used in two of their speaker systems: "Woofer speed @ 50 Hz: 6.3 ms (comparable to most 8" drivers)."—Ed.]

I'm no physics major, but isn't speed measured in distance divided by time? Well, they got a time; I wonder what the distance was? I can see myself asking the cop who pulled me over how fast I was going, and he says, "Sir, you were going 3 seconds."

There's another thing; they say that this woofer speed is "comparable to most 8" drivers." I looked through all my driver catalogs and couldn't find one 8" driver with a 6.3 ms speed rating. Go figure....

Sincerely,
Steven A. Crosby
APO AE

A "fast woofer" is one of the techno-illiteracies of the high-end audio ghetto. As I've said more than a few times, if a woofer were fast, it would be a tweeter. Tweeters have a fast rise time, another way of saying extended bandwidth. Woofers don't have that and don't need it. Woofers need to be well-damped, meaning that they must shut up quickly after the signal stops. I think that's what the tweaks really mean by "fast" in this context.

That Martin-Logan spec is probably some kind of settling time after some kind of specified pulse excitation—I'm only guessing. The amplitude information is missing, as you point out. Martin-Logan is known to me as a serious, engineering-oriented loudspeaker company, not a tweako cultist operation, but in this case there appears to be a bit of terminological pandering to the audio-salon cowboys. The good news is that the latest Martin-Logan literature I picked up at the Summer CES no longer includes that questionable spec. Maybe others like you got to them.

—Ed.

The Audio Critic:

...Concerning the ACI G3 review in Issue No. 19: "floor bounce" is the amplitude variation with frequency due to different path lengths of the direct radiation from the woofer and the reflected radiation from the floor. Frequencies affected vary with the height of the woofer and the listener's location. The "Allison Effect" is a variation in woofer power output due to interference with reflections from nearby room boundaries. The frequencies affected depend on woofer-to-boundary path length. These two effects are not the same.

Sincerely,
Robert T. Kuntz
Medford, OR

The "Allison Effect" was discussed in detail by Roy Allison himself in his interview with David Ranada (Issue No. 18, pp. 54-55), so I'm not exactly a stranger to the concept. I plead guilty to having referred to it somewhat sloppily. Anyway, Roy doesn't like the woofer to be mounted high up—right?—and in the ACI G3 it isn't. That's basically what I was trying to say; your commentary is more precise.

—Ed.

The Audio Critic:

So far I've only read two issues of your magazine. It has confused me, angered me, made me think, and thoroughly delighted me. Thank you....

...Can you please explain to me why the high-end mags consider the Toslink output inadequate? I Use it to clone CDs to DAT and wonder if I'm compromising quality.

Thanks for entertaining and enlightening me.

Sincerely,
Patrick T. Chamberlain
Albuquerque, NM

The theoretical argument against Toslink is that the bandwidth of the plastic fiber-optic interface may not be quite adequate, resulting in imprecise recovery of the clock and consequent jitter. But, if there is jitter, it will inevitably show up in the THD + N versus frequency measurements at the higher frequencies, and I have never seen a difference in that test between Toslink and coax. Furthermore, in digital-to-digital copying such as you do, jitter is a total nonissue. So relax and stop worrying. Those 1's and 0's are not so fragile.

—Ed.

The Audio Critic:

...In Issue No. 18 you discuss line-level preamps. On page 18 you discuss capacitors in the signal path, with a reference to the Jung/Marsh articles in *Audio* magazine. I have enclosed a copy of a paper I wrote, entitled "Ceramic Capacitors," written in response to the Jung/Marsh articles. Jung & Marsh incorrectly assumed, based on the experiences with one specific type of ceramic capacitor (X7R dielectric), that *all* ceramic capacitors were inferior. My paper shows that there are several distinctly different types of ceramic capacitors, and only *some* of them are inferior. The COG/NPO dielectric is superior. A later paper by Jung takes a much more accurate view of ceramic capacitors.

Your discussion of capacitors compares the use of electrolytic capacitors in speaker crossovers with their use in preamps, where "...the voltage swings are much smaller and the loads larger by at least three orders of magnitude." Your point is that the smaller voltage swings and higher-impedance loads will mean that capacitor "...nonlinearities in the range of human hearing should not be measurable." If you look at the THD graph in my paper, you will see that THD of almost 3% is possible with the Y5V dielectric, with a load of 1 kΩ the high-pass filter. Even at a substantially higher-impedance load, distortion caused by the Y5V dielectric would still be very measurable.

I hope nobody would be stupid enough to use a Y5V ceramic capacitor in the frequency-determining network of an equalizer or electronic crossover, where it would be working in the audible frequency range. However, there may be numerous products that *do* use inferior types of ceramic capacitors for the op-amp feedback loops and/or frequency compensation networks....

Sincerely,
John W. Hardy
President
The John Hardy Company
Evanston, IL

David Rich, who wrote that sidebar on "Capacitors in the Signal Path," says he does not disagree with you, and I certainly do not. (But all dielectrics are guilty until proven innocent, aren' they?) In any event, designers of high-quality audio equipment almost invariably specify film capacitors for the smaller values at all points in the circuit where distortion could be an issue.

—Ed.

The Tweaks vs. the Pros: Is It a Bona Fide Debate Between Two Points of View?

By Peter Aczel
Editor and Publisher

The audio world has been conditioned to see this subject in a strangely distorted perspective. It appears that even some level-headed audiophiles could benefit from a bit of clarification.

When a controversy goes on too long, the issues tend to become distorted, sometimes beyond recognition. Originally, the hotly debated question in audio was whether or not good measurements guaranteed good sound. That was a legitimate debate, especially in those early days when measurement protocols were quite skimpy and the psychoacoustic correlations unexplored or unproven. Over the years, clouded by the agenda of the ultrahigh-end audio community, the debate degenerated into I-know-I-can-hear-it vs. prove-to-me-that-you-can-hear-it. The sad thing is that the unending back-and-forth on the constantly shifting ground of this controversy has left many audiophiles with the impression that a fundamental clash of philosophies was taking place, the audio world's equivalent of capitalism vs. socialism, religion vs. atheism, Republicans vs. Democrats, protectionism vs. free trade, batting pitchers vs. designated hitters, etc., etc. That is an enormous misperception. The way the lines are drawn today, the debate doesn't have two arguable sides. It's more like laetrile vs. the AMA or the Ku Klux Klan vs. civil rights.

Consider the typical equipment reviewer for one of the "subjective" audio journals. He receives a new amplifier, plugs it into his system, listens to it for a while, and then declares that it has better soundstaging but a less liquid upper midrange than the amplifier that was in the system last week. No side-by-side comparison at matched levels, no attempt at blind listening, just total confidence in his golden ears, his exquisite judgment, and his perfect aural memory. Isn't that the height of narcissistic presumption (not to mention psychoacoustic illiteracy)? And isn't it the height of mindless credulity for an audiophile to follow such a reviewer's advice? There's no plausible side-taking in a two-sided philosophical controversy there. That reviewer is simply an unprofessional slob who can't be bothered with time-consuming homework, and the audiophile who relies on him for guidance is an undiscriminating innocent. (Of course, the mystic rituals of high-end audio have always attracted the natural-born cultist, and cultists are prone to believe anything, on any subject. To bring up an extreme example—who would have thought that the third-rate con man, child molester, and dilettante rock 'n' roller who called himself David Koresh would be believable as the Son of God to a group of outwardly nonpsychotic persons? Next to that, the tweakiest audio reviewer is a paragon of credibility.)

Please note that I am not minimizing the importance of listening. Audio is *about* listening, and ultimately every piece of audio equipment stands or falls on its performance in the listening room. Listening is an all-important part of equipment evaluation but it must be a great deal more structured and disciplined than the totally chaotic, self-indulgent, freeform exercise it has become in tweako reviewer circles. Yes, you can enjoy music without critically listening for audio quality—just go with the flow, man, etc.—but no, you can't just dip into amplifier A on Tuesday and amplifier B on Wednesday like a restaurant reviewer and then declare A to be superior to B. The ground rules of meaningful listening tests were explained in considerable detail in Issue No. 16, so I don't intend to rehash them here; what I want to convey in this discussion is the absence of even marginally defensible arguments on the tweaks' side of the debate—it is in effect a nondebate, with all the informed opinion on the other side. Indeed, it is an essential requirement for obtaining membership in the tweaks' circle to have no university degrees in engineering or physics and to have no professional status in audio anywhere outside the high-end manufacturing/retailing/writing ghetto. If there are any exceptions to that, I'm not aware of them.

Let us examine some of the favorite tenets of the tweaks and see whether a scientifically respectable two-sided argument can be occasioned by any of them.

Listening for differences.

Maybe we should start with the basic shibboleth that takes the form of "why don't you just listen for yourself and you'll hear the huge difference between the Mark Levinson and the Bryston" (to pick an ultrahigh-end and a merely high-end brand at random). You can listen for yourself till the cows come home and you won't be able to reach a valid conclusion unless (1) you can readily switch between the two units side by side, as quickly or slowly as you wish, (2) their identities are concealed from you when you try to determine from their sound which is which, and (3) their levels are matched within ±0.1 dB. All psychoacousticians, psychophysicists, statisticians, and credentialed audio electronics authorities agree with that statement; only the Atkinson/Harley crowd opposes it with all kinds of untutored aesthetical, psychological, and philosophical babble without proof. I don't consider it to be an intellectually respectable debate because there aren't respectable practitioners on both sides of it. Casual, sequential, "open-loop" listening evaluations are pretty close to worthless; nobody with serious credentials defends them.

That pesky level matching within ±0.1 dB is a key issue. I have found that tweako magazine reviewers as well as private experimenters tend not to do it even when they pay lip service to it; often they lie about having done it because it is too damn time-consuming, boring, and irritating to do accurately. You need a digital voltmeter with dB display, a stereo attenuator with coarse and fine adjustment, a reliable sine-wave signal source, and lots of patience—because four channels must be matched in a stereo comparison, sometimes with temperamental balance controls, etc., in the signal path. Not for the short-attention-span, instant-gratification, yeah-that's-it audiophile. But it must be done without fail, otherwise all bets are off. A mismatch of 0.3 dB is definitely audible and will most probably be perceived as a difference in quality rather than level. (An easily identifiable level difference is usually closer to 1 dB.) As soon as A sounds even slightly different from B, audiophiles will declare one to be vastly superior to the other, and that's how the B.S. starts.

My most shattering experience in audio was the first time I bothered to match levels accurately. The two preamps I was testing began to sound exactly the same, and my audiophile belief system crumbled. (Yes, they still would have sounded different if they had differed by more than 0.2 or 0.3 dB in frequency response, but they were both dead flat, alas.) It has become my firm conviction that level matching is the big stumbling block that prevents rational audiophiles from leaving the tweako camp.

Let me hasten to add (although I shouldn't really have to) that even under properly controlled listening conditions numerous audio components will sound different and therefore require subjective evaluation. Those who are able to read *The Audio Critic* without moving their lips know very well that the "everything sounds the same" label the various tweako journals try to stick to our audio philosophy is a malicious misrepresentation. What we insist on is that there are no *unexplainable* sonic differences, no mysterious "X factor" that golden-eared high-end designers dial into their creations. Audible differences are due to frequency response, distortion, noise, impedance effects, and so forth—in other words, to quantifiable phenomena. In most cases the quantification is easy, in some not so easy, but the mechanism whereby the audible difference occurs is never inexplicable. Once again, try to find a genuinely credentialed authority who disagrees with that.

The glowing bulb fallacy.

It really bugs me that so many audiophiles still insist that "tubes sound better." Such nonsense! At the risk of sounding repetitious, I challenge all you tube freaks to find someone with a university graduate degree in electronics who is of the opinion that there are tube circuits for audio applications whose ouput, with any given input, cannot be totally duplicated by a solid-state circuit. Come on, guys, your vacuum-tube gurus and designer heroes are all dilettantes—former audio salesmen, sales managers, tweako audiophiles turned manufacturers, anything but credentialed professionals in electronics. Show me a published IEEE or AES or other *scientific* paper documenting the superiority of vacuum tubes over properly operated, modern solid-state devices in audio circuits. There aren't any. Get real, guys.

Yes, vacuum tubes are great for broadcasting applications, outputting kilowatts of RF power. They're also good for generating various pleasant, "musical" distortions in guitar amplifiers, compressors, equalizers, etc., but of course solid-state circuits can be built to mimic those same, nice, old-fashioned distortions without requiring perishable parts made in disintegrating Eastern European countries. Some audiophiles love that kind of distortion in their main power amplifier and preamplifier as well, but then why not get an Aphex Aural Exciter or some other external grunge box? That you can at least turn off. (See page 55 of Issue No. 16 to understand what a tube power amplifier with a high output impedance—i.e., low damping factor—can do to the frequency response at the speaker terminals. Sure you can hear a 2 dB range of response fluctuations; you're even allowed to love it; but I can give you the same thing with a 10-watt 1-ohm series resistor.)

Best-case scenario: somebody comes up with a tube amplifier or preamplifier that does everything the best solid-state units can do, except for an order of magnitude higher distortion—mostly second harmonic—plus a higher noise floor and significantly shorter mean time between failures. That's the absolute best you can hope for, so why bother?

The magic cable fallacy.

At the Summer CES in Chicago, Ray Kimber of Kimber Kable walked around with a length of speaker cable (I think it was about four feet long), telling everybody—and relishing the shock effect—that this latest-and-greatest Kimber product would be selling for $15,000 the pair. I asked him whether it sounded better than no cable at all, with the amplifier terminals soldered directly to the speaker terminals—you could do that for a few dollars with, say, two inches of copper bar per terminal—and he said no, how could it, but it sounds just as good! Then I asked him whether he believed he could hear the difference between his magic cable and some kind of cheap cable I proposed to cobble together, having exactly the same resistance, same inductance, and same capacitance as his. He said yes, of course, 100% of the time. I challenged him to a double-blind listening test to prove that, and he accepted—but didn't stick around to complete the arrangements. (Don't hold your breath, dear reader.)

My point is that any cable, whether it costs $1.50 or $15,000, is an RLC circuit and will behave accordingly. Whatever effect the cable has on the sound, that effect will be due *exclusively* to the R, the L, and the C of the cable, interacting with the output impedance of the amplifier and the impedance characteristics of the speaker. To make R, L, and C as small as possible—and thus have minimal effect—costs *some* money but not many hundreds, and certainly not thousands, of dollars. Super-expensive cable does absolutely nothing for you. I explored this subject in some detail in Issue No. 16 and cited as my main supporting authority the writings of Richard A. Greiner, Ph.D., an E.E. professor at the University of Wisconsin. Since then another professional engineer, Fred E. Davis, has added his voice to those of the advocates of science and reason on this subject, in the form of heavily documented articles in the *Journal of the Audio Engineering Society* (June 1991) and *Audio* (July 1993). He basically comes out in favor of reasonably low R and the lowest practicable L, and observes that the differences are small in any event. The voices on the tweako side of the issue, now as before, belong to lightweights without any comparable credentials. Again, no genuine debate.

The antidigital fallacy.

Digital audio has brought us flatter frequency response, deeper bass, wider dynamic range, lower THD, lower noise floor, safer storage, and greater editability than any other technology in the history of sound reproduction. It has every scientific reason to sound better than analog, and it does—the possible exception being 30-ips analog tape with Dolby SR, which is capable of sounding equally good (with some qualifications), but which the tweaks aren't even talking about. They talk about vinyl, for crying out loud; they say digital just doesn't have the same airiness, smoothness, front-to-back depth, dimensionality, tonal gradations, etc., etc. This is truly sickening drivel, without any foundation in logical thinking or accurate hands-on observation.

Vinyl is not a primary medium; it is nearly always a transfer from tape, sometimes even digital tape, except for the very few direct-to-disc recordings. The process of transferring tape-recorded material to vinyl entails measurable losses and distortions; reading the vinyl groove with a pickup entails further measurable losses and distortions, not to mention mechanical ticks, pops, and swishes. The tweaks appear to like the results of this flawed process, especially the added L-R component introduced by the insufficiently orthogonal motional characteristics of the cutter and pickup. That's your extra airiness and depth, tweaks—and it isn't on the *analog* master tape! Nor is the lovely "smoothing" effect of the vinyl noise. At the very least, the comparison of analog vs. digital should be master tape vs. master tape—but it never is because the digitophobic cultists don't know any better. (Actually, the digital recording in the comparison could be a CD or an R-DAT because the codes are exactly the same as on the master tape, but the cultists are hazy on that too.)

Once again, where are the authorities in support of the antidigital arguments? At the tweako magazines and the tweako stores, that's where. And where are their credentials as authorities? Nowhere. Herbert von Karajan hailed the advent of digital recording with the remark that "all else is gaslight." But Harry Pearson didn't like it, nor did Michael Fremer. And now the great "digital expert" Robert Harley also travels cheerfuly with the analog-is-still-best crowd, although it's rumored he didn't even own any vinyl LPs back in 1989. Pearson, Fremer, Harley—versus Karajan? Versus John Eargle, Stanley Lipshitz, etc.? You call that a debate? Even the illustrious Edward Rothstein of *The New York Times* has been heavily qualifying his unconvincing antidigital quibbles lately. I think the more intelligent analog diehards are in retreat.

Making the l's and 0's more 1-ish and 0-ish.

The same element that bashes digital and extols analog seems intent on "improving" digital with nonsense products that you glue, paint, spray, rub, clamp, etc., on your CDs. This is so primitively unscientific that you can't even talk about a phony debate; here we are in witch-doctor territory. No double-blind listening tests have ever revealed improvements with theses devices; no authority with serious credentials has ever endorsed them. A CD stores numbers; you can either read a number or you can't—no massaging will make it "better."

Why not address problems that need solving? •

A New Look at Medium- and High-Priced Power Amplifiers

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

This will be an ongoing survey stretching over a number of issues. Seven units are reviewed in this first installment, which introduces a new method of measuring amplifier/speaker interaction.

Editor's Note: *Just like the CD player and preamplifier surveys by Dr. Rich, some of the writing that follows here is a bit more technical than many readers of general audio magazines may be prepared for. The overall evaluations and conclusions are loud and clear to anyone who has ever considered purchasing an amplifier, but some of the engineering explanations may be tough going for nontechies. I am not totally happy about that but I have not edited out any of the technical material because I feel that it should be in print and available for reference. Amplifiers have been the subject of incredibly stupid, ignorant, misleading writing in several publications that ought to know better, and we definitely need to add what we can to the small storehouse of authoritative amplifier information accessible to audiophiles.*

* * *

Introduction.

The design of a power amp is a nontrivial matter. A power amplifier is required to produce over 100 volts peak to peak. Low-impedance loads may demand peak currents in excess of 50 amps. The amplifier must be stable into an unknown load, which, depending on the speaker's design, can have a range of close to two orders of magnitude. The phase angle of the load can range from an almost pure capacitance to an almost pure inductance and everything in between. The amplifier must not blow up into an open circuit (when no speaker is connected) or a short circuit or when its inputs are hot-socketed or when the input level drives it into clipping.

If you had told me 20 years ago that commercially manufacturable power amplifiers would be designed to do all of the above and produce maximum distortion levels of 50 parts in a million, I would not have believed it. If anybody could have pulled off such a feat, I would have been sure the audio community would declare the designer a genius and his amp would be a most coveted item. As we will see below, such amps now exist, but their designers are not hailed; no, they are actually scorned by the high-end community. Anybody can make an amp that measures good, we are told. Instead, high-end reviewers and dealers will tell you a good-sounding amp is a work of art, not science. Do not worry about the numbers; they do not matter, you will be told. Do not expect the amplifier to be reliable, for this is also in conflict with good sound, they say. Well, I do not believe a word of this. In the reviews below and in coming issues I will recommend the best-built and best-performing amplifier. Amplifiers that come close but cost a lot less will also be recommended.

The sound, or the lack of it.

We still get letters asking why I do not discuss the sound of the equipment in my reviews. I do not know how *The Audio Critic* can be any clearer on this issue. If a piece of electronics has flat frequency response, vanishingly small static and dynamic nonlinearities, a high enough input impedance and low enough output impedance, a noise level below audibility, and high enough channel separation, then it is not going to have a sound. ABX testing confirms this. That is not to say all amplifiers sound the same. Clearly an amplifier can have insufficient voltage- and current-drive capabilities for a given loudspeaker, and that is going to be audible. Our test regime is designed to identify wimpy amplifiers. Amplifiers are also going to sound different if they do not satisfy the above conditions.

Take the much-praised Jadis JA-200 monoblock power amplifier, which sold for $17,500 the pair (four chassis, actually) the last time I looked. Here is a vacuum-tube amplifier with high static distortion, an inband slew-rate limitation, marginal stability, and high output impedance (low damping factor). Now this amplifier is going to sound *different.* The problem is that audio reviewers often confuse *different* with *better.* The Jadis is in effect a very expensive equalizer rather than a real amplifier. The Carver "t-mod" was basically an elegant experiment to prove the above statement and others like it. Instead of ending the discussion, the t-mod resulted in an attempt by the high-end community to discredit and destroy Mr. Carver. Despite all of the available evidence, the majority of audiophiles continue to want to *believe.* In what other field could the following quote, from a Bascom King review of the Jadis amplifier *{The Absolute Sound,* Issue 41, Spring 1986), be taken seriously?

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

Now, if this makes sense to you, you might as well stop reading this article, go out, and buy yourself a megabuck tube amplifier/equalizer. If the above quote does not make sense, then stick around—you are going to pay a lot less for an amplifier, and it is going to sound better than the Jadis. I hope you now understand why I do not need to characterize the sound of the individual amplifiers reviewed below. They all (except the Michael Yee special) meet the above electrical criteria for inaudible differences. We confirmed this with a series of ABX tests that included all of the amplifiers. (All except the Michael Yee unit, which blew up—twice! Remember what the dealer told you: if it sounds good it's going to be temperamental, just like an Italian sports car....)

Why do they call it a power amp?

Let's start our technical discussion with the much-discussed fact that power output is not the parameter to characterize the drive performance of an amplifier. All commercial amplifiers are voltage-drive. (Papers on current-drive amplifiers have been presented in the AES *Journal.)* Ohm's law requires that when the amplifier imposes a voltage across the load, the amplifier must also be able to source the required current. For the case of a resistive load, a one-to-one correspondence exists between the voltage output of the amplifier, the current flow of the amplifier, and the power dissipated by the load. When an amplifier is connected to a purely reactive load, the voltage and current waveforms are displaced by 90°. Under these conditions, no power is dissipated by the load, but the amplifier must still drive the load with the appropriate voltage and it must source and sink the current from the load. Indeed, the power being dissipated by the amplifier itself actually increases when driving reactive loads [Benjamin 1992]. It is thus more appropriate to characterize the amplifier in terms of the voltage it can supply across a variety of load conditions. Both the magnitude of the impedance of the load and the phase angle of the impedance need to be varied.

When an amplifier is required to source (or sink) a significant amount of current, losses in the power-supply transformer and the output transistors reduce the amount of voltage which can be imposed across the load. An amplifier designer is presented with an interesting tradeoff. For a given size and cost, a transformer can be specified which supplies a large unloaded voltage but which has a large internal resistive loss. Conversely, a transformer can be chosen with a lower unloaded voltage but a smaller internal resistive loss. When driving a high-impedance resistive load, the larger voltage available from the first kind of transformer will allow more power to be delivered to the load. When the load is reduced in value, the transformer losses become significant and the power-supply voltage (and hence the available output voltage) decreases, thus less power can be delivered to the load. This is where the second kind of transformer becomes more appropriate. Although its unloaded voltage is lower, its voltage under these load conditions is higher, and the power to the load is thus also higher.

Even if a lossless transformer existed, current output would be limited by output transistor losses. Adding additional transistors in parallel reduces these losses and allows the amplifier to produce a more constant voltage output as the load is varied. In addition to more output transistors, more heat sinks must be added to the amplifier if the amplifier is to drive a low-impedance load. This is because amplifiers have a finite conversion efficiency. Not all the power reaches the load. Assuming a resistive load and a lossless output device, a class A amplifier is only 25% efficient! An ideal class B amplifier is only 78.6% efficient. Since more power is being transferred to the low-impedance resistive load, more power is being dissipated by the amplifier and more heat must be dissipated. (Remember, reactive loads can make this situation worse.) An additional factor to be considered is the size of the primary filter capacitors. The filter capacitors hold the power rails up in between conduction cycles of the diode bridge connected to the power transformer. The capacitors must be sized large enough so that the power supply does not sag under maximum current draw. An amplifier that can supply a ±140 V peak-to-peak swing into a 1-ohm load continuously is putting 2400 W into a resistive load (into 8 ohms only 300 W is being delivered to the load). The Krell KSA-300S is claimed to be such an amplifier; it weighs 185 pounds, has a 5 kVA transformer, 0.27 F of supply capacitance, and costs $8500. Now, if you are driving a pair of 1-ohm loads continuously in stereo, you need this amp. If not, this is clearly overkill. That does not mean you should not spend the money; just look at the expense in the same light as spending $80,000 for a sports car that can go 180 mph.

Measurements and current limiters.

Even if a power amp cannot source high values of current on a continuous basis—because of transformer losses, inadequate heat-sink sizing, and high output-device drops (leading to rapid internal heating)—it may be able to do so for a short period of time. This is accomplished by oversizing the main filter capacitors so that they keep the power rails up for a short period of time, even though the losses in the transformer would prevent the rail from being held at a high voltage continuously. Output-current-limiting circuits or fuses limit the time during which the output devices source current beyond the maximum allowed under continuous operation. These same elements prevent the amplifier from being damaged during fault conditions, such as a short across the amplifier. That extra headroom available for a short duration is called dynamic power. *The Audio Critic* is the first magazine to use a new method for assessing dynamic power output, called *The PowerCube.* The PowerCube, a software-driven automated instrument with a PC front end, measures the maximum voltage the amplifier can deliver to the load at 1% distortion for a duration of 20 ms at 1 kHz. The loads have impedance magnitudes of 8, 4, 2, and 1 ohm, and phase angles of -60°, -30°, 0°, +30°, and +60°. The graphic output of the instrument shows the 20 data points connected to form a more or less cubelike polyhedron, which illustrates at a glance the true dynamic performance of the amplifier. The test takes only a few minutes per channel. Under ideal conditions the voltage sourced by the amplifier should be a constant, and The PowerCube should look like Figure 1. We haven't seen one like that so far. A real amplifier, even a very good one, is going to exhibit some tilt in the cube, as shown in Figure 2. The extent of the tilt depends principally on the primary capacitor reservoir size, the losses across the output devices, and the action of the current limiter if it is present.

A brief detour into the operation of the current limiter is required at this point. The current limiter must consider the voltage across the device, the current flowing in the device, the temperature of the device, and the amount of time that a high-current condition has existed. Ideally A/D converters would transmit this information to a microcontroller, which would compare the operating conditions to a template of the SOA (safe operating area) conditions for the output devices and determine if the output current should be limited or if the amplifier should be shut down. Krell literature hints at such an approach in their very expensive amplifiers. The next best thing is to use analog circuitry to calculate the instantaneous power dissipated by the devices. Additional circuits integrate this value and then determine if it exceeds the SOA of the devices [Didden 1983]. In most amplifiers a much simpler one-transistor circuit is used to limit the current. The base-emitter junction of the transistor is placed across resistors in series with the output devices. When the current in these resistors becomes large enough, the transistor is turned on. The collector of the transistor is connected to the base of the output device or sometimes to an earlier part of the circuit. When the protection transistor is on, current is diverted away from the base of the output device, reducing the current the output device can supply. Some additional circuitry may be included to add a time delay to the circuit's action or suppress oscillations that could occur when the limiter circuit is tripped [Leach 1980]. Properly designed, the circuit can work reasonably well, but several researchers have noted major problems with the circuit if it is not optimally designed [Holman 1981], [Baxandall 1988], [Fairwood and Reed 1991]. When the current limiter is not designed properly, it will typically limit current prematurely or will limit when the load is reactive. Although it is possible for a poorly designed current limiter to trip too early on reactive loads under some conditions, it is necessary for the current limiter to activate at lower current levels when the amplifier is driving a reactive load [Benjamin 1993] if the output stage is underdesigned.

The PowerCube helps us assess the performance of the current limiter from the tilt of the cube and from a loss of voltage into reactive loads. There will be a ridge through the center line of the cube and the top surface of the cube will bend downwards away from that center line if the current limiter is malfunctioning (Figure 4). Another thing The PowerCube can tell us is how stable the amplifier is into reactive loads. If the cube has a missing corner (Figure 5), it is likely that the amplifier is unstable or marginally stable into the load. When unstable, the amplifier distorts at a lower power level, or oscillations start to occur, limiting power output. You may ask if it is important for an amplifier to work into a 2- or 1-ohm load. Surprisingly, the answer is that many popular loudspeakers can present difficult loads that require this [Vanderkooy and Lipshitz 1986], [Otala 1987], [Baxandall 1988], [Fairwood 1991], [Benjamin 1992]. Note that The PowerCube and the theoretical work behind it are relatively recent developments. An amplifier which fails the PowerCube test is going to sound different when driving a loudspeaker that constitutes a complex load. Tube amplifiers fold over and die on the PowerCube test, owing to their impedance-matching transformer. A final PowerCube test is the peak current that the amplifier can source and sink. This is measured with one cycle of a 10 kHz signal into a 0.1-ohm load (a virtual short circuit).

I devoted a good deal of space to the characterization of linearity errors in electronics in my introduction to the preamp survey in Issue No. 18. I will not repeat all that here, but I will summarize a couple of important points. Frequency-independent nonlinearities result in static distortion. Linearity errors which are de-

Typical Examples of Amplifier Measurements with *The PowerCube*

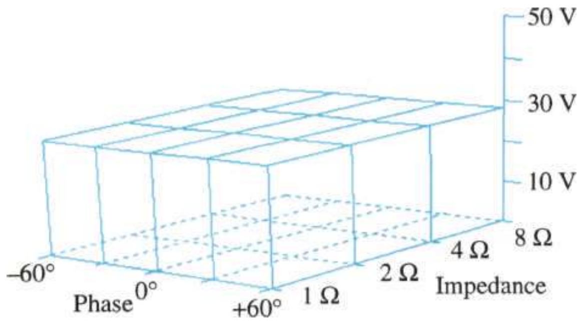

Figure 1: The perfect amplifier that never existed.

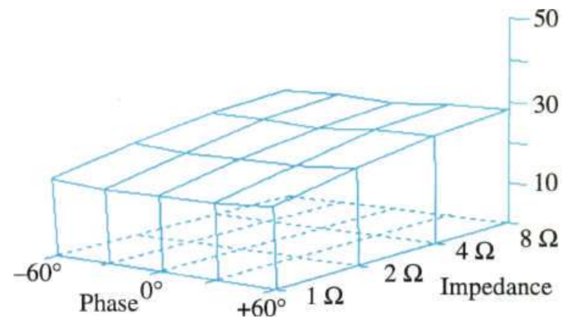

Figure 2: An amplifier with a good power supply.

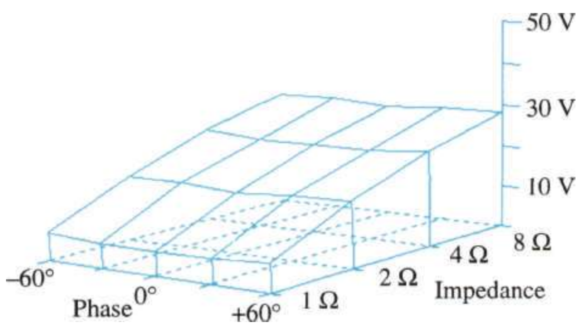

Figure 3: An amplifier with a poor power supply.

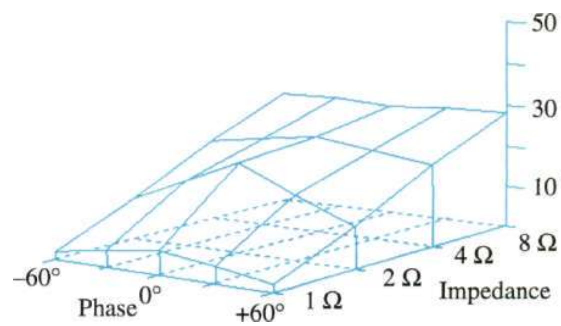

Figure 4: Very badly designed current limiting.

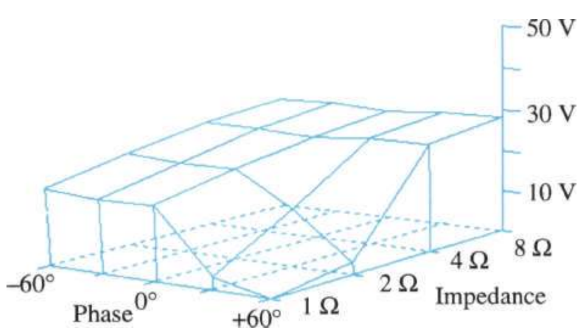

Figure 5: Severe oscillation with a reactive load.

The PowerCube seeks a target value of 1% THD at 20 data points. It uses 1 kHz sine-wave bursts of 20 ms duration. In a separate test it also measures instantaneous peak current (one cycle of 10 kHz into 0.1 Ω).

pendent on frequency are categorized as dynamic linearity errors [Borbely 1989]. Dynamic errors are not more important than static errors. An amplifier which has a steady -60 dB harmonic distortion across the frequency band is not better than an amplifier which has -100 dB distortion at 20 Hz and -80 dB distortion at 20 kHz.

We use THD tests exclusively to characterize amplifier nonlinearity. Human hearing is not very sensitive to low-order harmonic distortion, but there is greater sensitivity to nonharmonically related tones, such as those generated as intermodulation products in distortion tests using two or more tones. We do not perform such separate IM distortion tests here because they characterize the same nonlinearities identified by the THD tests. A nonlinearity that gives rise to a high 20 kHz THD will also cause inband distortion products in a multitone test. A full-scale 20 kHz test has the advantage that it has the maximum rate of change of any inband test signal and it characterizes both even- and odd-order nonlinearities [Borbely 1989], [Jung 1979]. Transient intermodulation effects [Otala 1970] are also covered in this test. It has the disadvantage that the distortion components measured to characterize the nonlinearity are out-of-band. In Issue No. 18 I pointed out that feedback rates cannot be increased to reduce 20 kHz distortion as they can be at lower frequencies because this will lead to stability problems. Open-loop bandwidth and open-loop distortion play an important role in determining the closed-loop 20 kHz distortion of an amplifier [Marsh 1985]. As I stated previously in Issue No. 18, it can be shown using stochastic processes that a nondeterministic music signal responds to a nonlinearity in the same manner as a static sine wave. The audiophile folklore that sine-wave tests do not fully characterize the nonlinearity of an amplifier is just plain wrong. As stated in Issue No. 18, we do not perform exotic tests such as phase intermodulation distortion because these tests have been shown to be of questionable value [Cherry 1982], [Cordell 1983]. In a power amplifier the distortion measurements are made with the amplifier under load. We performed the distortion tests under both 8- and 4-ohm loads. None of the amplifiers reviewed below is designed to source continuous power into a load of less than 2 ohms, so we did not test them for THD below 4 ohms.

Damping factor is a specification unique to audio power amplifiers. It is in effect a measurement of the output impedance of an amplifier, being the ratio of an 8-ohm load to the amplifier's output impedance. In a modern amplifier, damping factor at lower frequencies is often greater than 1000. Cable resistance will thus dominate. At high frequencies the damping factor may decrease. This can be due to reduced feedback rates in the amplifier, which are the result of dominant-pole compensation or feed-forward compensation [Leach 1988]. It is more likely, however, to be caused by the presence of a series inductor at the output of the amplifier, outside the feedback loop. The purpose of this inductor is to decouple the output stage from a capacitive load. The capacitor would otherwise create a pole in the amplifier's transfer function, which could lead to poor stability margins or oscillations. Your speaker cable provides series inductance free, so this will dominate an amplifier which does not have a series inductor. Please note that tube amplifiers can have very low damping factors. When a tube amplifier with a low damping factor is connected to a high-capacitance (read $100-per-foot) speaker cable and a difficult speaker load, the frequency response at the speaker terminals is anything but flat (see your Editor's article on speaker cable in Issue No. 16). Once again we see that tube amplifiers can sound different—different, not better.

MOSFETs vs. bipolar transistors.

Serious designers who believe that MOSFETs are a better choice for the output stage of a power amplifier are in a minority, but important thinkers such as Robert Cordell [Cordell 1984] and Erno Borbely [Borbely 1982], [Borbely 1984] are among their ranks. Let's first dispose of the tweako cultist dogma. The cultists say MOSFETs perform more like tubes because they are voltage-drive devices. Tubes have tremendous disadvantages and no advantages in a power amp, so this argument can be dismissed without comment. MOSFETs have significantly higher transconductance (the change in drain current for a change in gate-to-source voltage) than tubes (but lower than a bipolar—see below); thus they are usable in an output stage without a transformer. MOSFETs, unlike tubes, are available in complementary pairs.

Next, cultists will tell you that MOSFETs are more linear than bipolar devices. It is true that an unloaded and undegenerated common-drain MOSFET stage driven with a voltage source is more linear than an unloaded and undegenerated common-emitter stage driven with a voltage source, but no serious designer would use an undegenerated common-emitter amplifier in an output stage. With proper degeneration the bipolar stage will exhibit less distortion than an equivalent stage using MOSFETs. One reason for this is the fact that for a given bias current the tranconductance of a bipolar will be higher than of a MOSFET. Another reason is that the nonlinear emitter-collector output resistance of a bipolar device is much lower than the source-drain output resistance of a MOSFET. The open-loop output resistance of a follower stage is directly proportional to transconductance. The resulting higher output resistance of a MOSFET follower stage can lead to degradation in stability [Leach 1988] as well as lower damping factors.

If a MOSFET is used in a source-follower configuration, it will have significantly higher distortion than a bipolar device, owing to its lower transconductance and the fact that its transconductance varies considerably as the drain current changes. As the output voltage increases across the load, the output device must sink more current and the voltage between the gate and source of the MOSFET must increase in order to supply the current. The amount of this change is dependent on the device's tranconductance. The value of the transconductance increases as the current through the device increases. Thus when the output is near ground, the output device is sinking only a small current. The amount of gate-to-source voltage change required to move the output 1 V will be higher than when the output is close to the supply rail and the device is sinking significant current. This nonlinear change in gate-to-source voltage with a constant change in output voltage leads to distortion of the signal. Since bipolar devices have a much higher tranconductance, the nonlinear change in the base-to-emitter voltage over output current variations is small, and less distortion results.

The large variation in transconductance at low drain currents in MOSFETs leads to significant crossover distortion, since the total tranconductance of the output stage is reduced near the center ground level where current in the output stage is low. Overall negative feedback is not effective in reducing high-frequency crossover-notch distortion [Cordell 1984]. Cordell has demonstrated that the distortion of a MOSFET output stage can be reduced using an output error correction technique developed by Hawksford. Cordell produced a 50-watt MOSFET amplifier using the Hawksford technique and obtained a 20 kHz distortion figure of-105 dB. Caution must be used with this measurement, since it represents the result from one low-powered prototype. It does suggest that a scaled-up commercial version of this amplifier could be a world-beater, but unfortunately nobody has ever produced a commercial version of the Cordell design.

MOSFET devices have higher on resistance than bipolar devices because of the lightly doped drain region of the MOSFETs, which is required so that the devices will have high breakdown voltages. This reduces efficiency and increases thermal dissipation. A new device called an IGBT has recently been introduced, which has lower output resistance for a given die size. In other respects an IGBT is similar to a power MOSFET and is manufactured with a modified MOSFET process.

The total voltage required to keep a transistor turned on is called its forward bias. The high forward bias of MOSFETs trades away power-supply headroom and increases power dissipation, which must be addressed by adding additional heat sinks. The problem gets worse as a MOSFET is required to source current. Because of its lower transconductance than that of a bipolar device, the additional voltage drop from the gate to the source becomes much larger when driving significant current, and this further limits headroom. The principal parameter that sets the required forward bias voltage of a MOSFET is the threshold voltage. Bipolar devices have no parameter equivalent to the threshold voltage. MOSFETs are harder to parallel than bipolar devices because of the large variation in MOSFET threshold voltages. Complementary pairs of MOSFETs are less well-matched than bipolar devices, resulting in less cancellation of even-order harmonics in a push-pull stage. This is because MOSFETs are unipolar majority-carrier devices. They are either *n*-channel or *p*-channel. The *n*-channel MOSFETs operate with electrons; the *p*-channel MOSFETs use holes. On the other hand, *npn* and *pnp* devices use both (majority and minority carriers); that's why they are called bipolar.

A disadvantage of bipolar devices is related to the minority carriers. In the active region, bipolars accumulate charge in the base. When an output circuit makes the transition to class B, one of the output devices should shut off. The turnoff time of the device is delayed until the minority carriers can be cleared. This can lead to increased distortion in the crossover region. Designing a robust predriver stage, so that the minority carriers can be quickly removed, reduces the distortion effect. Best results are often achieved if all the predriver stages are biased to operate in the class A mode. Dynamic output-stage biasing circuits can also be used to eliminate the distortion due to minority carriers, by preventing any output-stage device from turning off completely. This scheme is often called sliding class A biasing. The current in the output devices is monitored and if it falls below a preset level (at the point where the circuit is going to go into the class B region), the bias voltage across the output stage is increased. This prevents any output devices from turning off. The dynamic bias circuit is a nested feedback loop in the amplifier. Problems associated with making sure the loop is wideband enough for the application and assuring it is stable must be addressed. To vary the output stage's bias voltage, the bias parameters of the second gain stage must be varied. This can degrade performance of the second gain stage. For these reasons, dynamic bias has apparently not been as effective as had been hoped and the technique is less popular today than several years ago.

The collector current of a bipolar transistor has a positive temperature for a fixed base-emitter voltage. Thus, as the device gets hotter, it draws more current and gets still hotter. If the process is allowed to continue, the device can draw enough current to destroy itself. This process is called thermal runaway. To keep the quiescent current of a bipolar output stage stable as the heat-sink temperature changes, some parts of the bias network that sets the quiescent current are mounted on the heat sink to provide thermal feedback. This thermal feedback cannot respond instantaneously to heating at the power transistor's junction; thus it takes a relatively long time for the bias to be stabilized, and bias currents can vary significantly as signal conditions vary. Thermal settling components can be a source of distortion. In addition to a thermal-tracking bias circuit, an overtemperature sensor is typically mounted on the heat sink. If the heat sink goes above a predetermined temperature, the amplifier is shut down.

Now, MOSFETs are not as sensitive to temperature changes as bipolar devices. MOSFETs are often claimed to have a negative temperature coefficient of drain current. While it is true that the current gain of a MOSFET decreases with temperature, the threshold voltage also decreases with temperature. Since the current flow through a MOSFET increases as the threshold voltage decreases, this component of the current change is positive with temperature. The actual direction of the temperature coefficient is thus dependent on the way the MOSFET is biased. In any case, the smaller value of the temperature coefficient means that MOSFETs do not require a thermal feedback circuit to stabilize the bias, or they require only a small amount of feedback. As a result MOSFET output stages show much smaller amounts of thermal distortion. Didden uses a circuit which senses the average quiescent current of the output stage and compares it with a reference value [Didden 1983]. The circuit then changes the bias voltage if the current is not at the correct value. This improves thermal stability, since the current of the output stage is being monitored and controlled directly.

A related phenomenon of bipolar transistors is called secondary breakdown. Secondary breakdown occurs when a local hot spot develops on the surface of the device as a result of collector current concentrating on small areas of the silicon substrate. Current conduction becomes nonuniform and is concentrated at these hot spots. This leads to further device heating and eventual device destruction. The temperature of the case of the transistor will not show the presence of these hot spots that cause the device's destruction. The safe operating area (SOA) of a bipolar is a set of voltage-versus-current regions as a function of the time for which the bipolar device will not enter secondary breakdown. Clearly the bipolar must be protected from entering these overload conditions. This protection is the current limiter discussed above. Properly designed, these protection circuits can make a bipolar amplifier very reliable. MOSFET output stages do not usually require current limiters; this allows for reduced production cost. When a local hot spot occurs in a MOSFET, the area tends to conduct less current, and this tends to equalize the temperature across the die.

MOSFET devices are often claimed to be faster than bipolar devices. Often these comparisons are of the large-signal switching characteristics of the device. Most power semiconductor devices are used for switching functions, not linear amplification. Under these conditions a MOSFET will usually win. Switching characteristics are important in the transition region of a class AB amplifier, as discussed above, but in a power amplifier device characteristics in the active region are most important. Here the winning technology is less apparent. One problem in making these comparisons is that state-of-the-art bipolar devices suitable for power amps are not readily available in the U.S. Most high-speed, high-power bipolar devices that are available in matched complementary pairs are manufactured in Japan. Advanced technology, such as ring emitters, is used in these devices. From the limited information I have on these devices, they appear to have band widths equivalent to those of MOSFETs. Very fast bipolar devices are available but they have breakdown voltages which are too low for a high-power amplifier. This can be worked around by placing a dynamic cascode device at the collector of the output bipolar device [Didden 1983]. As the emitter of the dynamic cascode moves with the signal level, it holds the V_{CE} of the high-speed output device to a low constant level, preventing breakdown.

Bipolar devices require base current to operate. One or more predrivers are required to supply this current, so that the second gain stage of the amplifier is not loaded down. MOSFETs do not require base current but they have very large input capacitance. Designing a wideband predriver to drive this load can be more difficult than designing the stage to interface current to a bipolar output stage. Fortunately, in a source-follower configuration the gate-to-source capacitor is bootstrapped, reducing the effective input capacitance of a MOSFET-follower output stage.

Going to the head of the class.

It is strange that tweaks will dismiss most causes of static linearity as unimportant, but they are obsessive about the class of operation of the output stage. In a true class A amplifier the quiescent current is set large enough so that it can drive the maximum current demanded by the load [Gray and Meyer 1984]. In such a configuration the amplifier is dissipating more power under no-load conditions than when it is putting out maximum power. Snyder provides a useful distinction between class A and class AB amplifiers [Snyder 1990]. A class A amplifier is limited by its quiescent current, and a class AB amplifier is limited by the output voltage limits. We have already discussed some problems with crossover distortion, which occurs in a class AB amplifier as the amplifier makes the transition from class A to class B, and have suggested potential solutions, such as dynamic biasing to keep all output devices turned on. One formal analysis of crossover distortion has been undertaken by Sandstrøm [Sandstrøm 1983]. He notes that in class A operation both devices are conducting, thus the open-loop output impedance of the amplifier is the sum of both of the transistors' tranconductances, and when the amplifier is in the class B mode, with only one transistor operating, the output resistance doubles. With a load on the amplifier, a three-section piecewise linear model of the transfer function of the output stage is constructed. In this model, the gain in the class B region is assumed to be a constant and smaller than the gain of the amplifier in the class A region. This simple model ignores the variation of transconductance with operating current, but it allows analysis. Sandstrøm's model shows that crossover distortion can be significant. Dynamic biasing will not help, since it keeps the side of the output stage not driving the load running at much lower current than the side driving the load. This side of the output stage has a much lower transconductance than the side driving the load, and the Sandstrøm model still applies. As explained above, crossover distortion can be difficult to reduce by the use of global feedback but can be reduced using the Hawksford circuit. Feed-forward circuits and current dumping topologies have also been proposed to reduce this crossover distortion. The interesting technique developed by Wurcer for the AD797 op-amp (see Issue No. 18, page 37) could also be considered, but it may not work outside a monolthic chip because of matching requirements.

Cherry has identified another distortion mechanism in class B amplifiers [Cherry 1981]. In an ideal class B amplifier the current flow into the output transistors is a half-wave-rectified sine wave. A half-wave-rectified sine wave contains high levels of harmonics. Mutual inductance between the power supply loop and the signal path can cause the power supply waveform to couple to the signal path, causing distortion. Careful attention to the layout of the amplifier is required to prevent this distortion mechanism from occurring. Snyder has proposed a balanced class A amplifier circuit which takes constant current from the supply [Snyder 1990]. In a balanced circuit the load is being driven at both sides. The current required by the two opposing output stages adds to a constant. Unfortunately this property does not hold for class AB balanced amplifiers. A balanced class AB amplifier can take advantage of a floating power supply for the output stage [Takahashi 1984]; thus signal-dependent current flow in the output stage is not transferred via ground to the preceding stages of the amplifier. Takahashi also shows in his paper that a balanced amplifier will have lower harmonic distortion independent of the class of its output stage .

A high-power class A amplifier is simply not practical, although some of these amplifiers have been produced commercially, because it is 25% efficient under load and dissipates full power as heat under no load. Fortunately, modern designers have been able to reduce the distortion mechanism in class AB amplifiers to a point where amplifiers can routinely produce THD-plus-noise levels below -80 dB.

The past and present.

Progress in the design of power amplifiers is directly tied to the progress in semiconductor engineering. Early solid-state power amplifiers were designed with unreliable germanium transistors. Designers unfamiliar with transistor circuit design used topologies similar to tube circuits. They even used transformers in the signal path. It is likely that some of the early transistor amplifiers sounded worse than the best tube competition. Silicon devices arrived in the middle '60s. Silicon power transistors were faster and more reliable. Before the development of modern devices, only *npn* power devices could be manufactured at reasonable cost. These devices were slow and had limited breakdown voltages. Output devices had to be placed in series to handle large voltage swings. Moving away from transformers, designers developed *npn-only* quasi-complementary output stages. These stages had asymmetrical performance parameters on the positive and negative swings of the output. Unipolar power supplies required output coupling capacitors, and the power supply rail was bootstrapped at the output of the second stage to improve output swing. Complaints about the sound of these amplifiers (they would measure lousy in our current test regime), the advent of fast complementary output devices, and the development of the two-gain-stage op-amp (741) topology brought forth a number of important design innovations in the '70s. Important contributions were produced by Bongiorno, Borbely, Carver, Cherry, Cordell, Curl, Garde, Greiner, Hawksford, Holman, Iverson, Jung, Leach, Meyer, Pass, Otala, and Takahashi among others. The result of their work was the modern solid-state amplifier. As a result of these researchers' innovations, the power amplifier became acoustically transparent, and tube amplifiers were hopelessly outclassed in *all* respects.

Table 1 shows the significant design elements of the power amplifiers in this survey. Other amplifiers in the table may be reviewed in future issues. I have also included kit designs from A-Train Ltd., Borbely Audio, New England Analog, and Old Colony Sound Lab. The Citation 22 which appears in the chart was reviewed in Issue No. 11. We wanted to include this excellent design in this survey but according to the manufacturer it will soon be replaced by a new model. We will be sent a sample of the new model when it becomes available. One extremely innovative design is the current-feedback amplifier of Mark Alexander [Alexander 1990]. Like the Cordell amplifier, it exists only as a low-power prototype, and no commercial design is available. It is *so* significantly different that it does not fit easily into the table. Current feedback, which was discussed in Issue No. 15, is now often used for *I*-to-*V* converters in CD players. Alexander shows that this feedback technique may have significant advantages in power amps also. He uses IGBTs in the output stage. Distortion of this amplifier is low. It happily produces 100 kHz square waves into 8-ohm loads. It is very stable into capacitive loads.

It can be seen from the table that most modern commercial transistor amplifiers are very similar to one other. Indeed, it is somewhat surprising that most of the

Table 1: Comparison of Significant Power Amplifier Topologies

	<——— *Differential Pair* ...>					***Buffer Stage or Compound 2nd Stage***	<———
MODEL	*Active Element*	*Comple-mentary Symmetry*	*Biased by Current Source*	*Cascode Stage*	*Load*		*Active Element*
A-Train Ltd. "Ampzilla III" (no longer offered)	Bipolar	Yes	Yes	No	Resistor	No	Bipolar
Acurus A250 $850.00	Bipolar	Yes	Yes	Yes	Resistor	No	Bipolar
Aragon 4004 MKII $1850.00	Bipolar	Yes	Yes	Yes	Resistor	Yes	Bipolar
B&K Sonata M-200 $998.00 each (mono)	Bipolar	No	Yes	Yes	Active	Yes	Bipolar
Borbely Audio (kit, made in Germany)	Bipolar	Yes	Yes	No	Resistor	Yes	Bipolar
Bryston 4B NRB $2195.00	Bipolar	Yes	No	No	Resistor	No	Bipolar
Citation 22 $1149.00	Bipolar	Yes	Yes	Yes	Resistor	Yes	Bipolar
Cordell (prototype) [Cordell 1984]	Bipolar	No	Yes	Yes	Active	Yes	Bipolar
Didden (prototype) [Didden 1983]	Bipolar	No	Yes	Yes	Resistor	No	Bipolar
Dynaco ST400II $995.00	Bipolar	Yes	Yes	No	Resistor	No	Bipolar
Hafler Series 9500 Transnova $1800.00	JFET	Self-biased	Yes	Yes	Resistor	No	Bipolar
New England Analog (plans only)	Bipolar	Yes	Yes	Yes	Resistor	Yes	Bipolar
PS Audio PS 100 Delta $1195.00	JFET	Yes	Yes	Yes (dy-namic bias)	Resistor	Yes	Bipolar
R.E. Designs LNPA 150 $2700.00 the pair (mono)	Bipolar	Yes	No	No	Resistor	No	Bipolar
Rotel RB-990BX $1100.00	Bipolar	Yes	Yes	No	Resistor	No	Bipolar
Sansui Vintage AU-X911DG $1250.00	Bipolar	Yes[6]	Yes	No	Resistor	No	Bipolar

[1]Nonlinear load implements soft clipping.
[2]The collector of the noninverting side of the differential pair is terminated into the emitter of the second gain stage. This is a folded-cascode-like topology.
[3]Part of a closed-loop feedback amplifier built around the output section.
[4]The Hawksford distortion correction circuit is used at the output stage.
[5]Dynamic output-bias-current set keeps quiescent current constant under different load conditions.
[6]Diamond Differential (X-cell) configuration biased with floating voltage sources.

(Not restricted to, nor including all, power amplifiers reviewed in this issue.)

——*Second Gain Stage*..................>			*Regulated Supplies on V Gain Stages*	*Output Predriver Stage(s) and Type*	*Number and Type of Output Devices*	*<—Coupling*	*Capacitors—>*	*Protection*
Cascode Stage	*Load*	*Push-Pull*				C_1	C_2	
No	Active	Yes	No	2 Bipolar	3 per side (balanced) Bipolar	Nonpolar electrolytic	No (DC servo)	A
No	Active	Yes	No	1 Bipolar	4 Bipolar	Electrolytic + film byp.	Electrolytic + film byp.	A, B, C, I
No	Active	Yes	No	1 Bipolar	4 Bipolar	No	NP 'lytic + film bypass	A,B,E
No	Active	No	No	None	1 MOSFET[1]	No	No (DC servo)	?
No[2]	Active	Yes	No	1 MOSFET	2 MOSFET	Film	No (DC servo)	A
No	Active	Yes	Yes	3 Bipolar[3]	4 Bipolar	Film	Electrolytic	B, D,I
No	Resistor	Yes	No	2 Bipolar	4 Bipolar	Electrolytic	No	B,F, I
Yes	Active	Yes	Yes	3 Bipolar	1 MOSFET[4]	Film	No	?
Yes	Resistor[1]	No	Yes (dynamic cascode on output stage)	2 Bipolar	4 Bipolar[5]	Film	No (DC servo)	H,I
No	Active	Yes	No	1 Bipolar	4 Bipolar	Electrolytic + film byp.	Electrolytic + film byp.	A, B, C, I
Yes	Output stage load 2nd stage	Yes	Yes	None	4 MOSFET	No	Electrolytic + film byp.	A
No	Resistor	Yes	Yes	1 Bipolar	4 Bipolar	No	No (DC servo)	D,I
No	Resistor	Yes	No	1 Bipolar	2 Bipolar	No	No	B,G,I
No[2]	Active	Yes	Yes (output stage also)	1 Bipolar	2 Bipolar	No	Electrolytic + film byp.	A
No	Active	Yes	No	2 Bipolar	5 Bipolar	Electrolytic	Electrolytic	A,B,I
No	Active	Yes	No	2 Bipolar	1 per side (balanced) Bipolar	No	No	E,F

Protection
A—DC rail fuses
B—Thermal sensing
C—Second-stage current limiting
D—Output-stage current limiting
E—DC input sensing
F—Excess input sensing
G—Output fuse in feedback loop
H—SOA monitor circuit (analog)
I—Output diodes to block inductive kick

NOTE:
The purpose of this table is simply the comparison of circuit features, not the establishment of criteria for evaluation. Please do not make the mistake of trying to derive a "figure of merit" for an amplifier from these data; that is not possible because the entries in the table do not necessarily represent right or wrong, better or worse, design solutions.

commercial amplifiers have topologies similar to that of Dan Meyer's 1973 Tigersaurus. This highly innovative amplifier, like the first commercial jet aircraft, the Comet, had major design weaknesses and, like the Comet, came to an early and catastrophic end. Note that none of the amplifiers in the table use sliding biasing, feedforward error correction, or current dumping. The Sansui is the only production amplifier in the table to have fully balanced outputs. Only the B&K and Hafler amps use MOSFETs. Below are the complete reviews of these amplifiers. Aspects of the design of the individual amplifiers are discussed in the individual reviews. References to capacitors in the signal path are based on the following basic schematic (see also Issue No. 18, page 18):

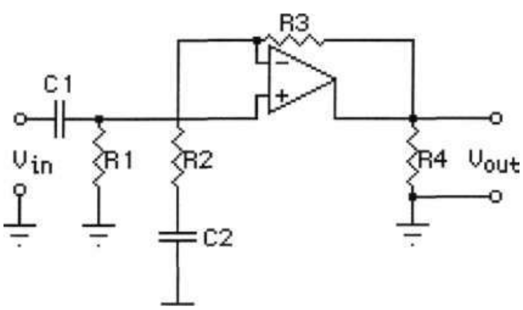

Figure 6: Capacitors in the signal path of a power amplifier.

Boulder 500AE

Boulder Amplifiers, Inc., 4850 Sterling Drive, Boulder, CO 80301. 500AE stereo power amplifier, $3999.00 (international version, $4099.00). Tested sample owned by The Audio Critic.

The original Boulder 500 was reviewed by the Editor in Issue No. 10. The 500AE is essentially the same amplifier minus the bells and whistles—one instead of 18 indicator lights, no input attenuators, balanced inputs only, no handles in the back. The sample we tested had been updated by Boulder in October 1990 and represents the current version.

This is the costliest amplifier so far in our survey, and it also turned out to be the best-performing amp within its power range in this group. I would love to tell you every detail on how they did it, but Boulder would not submit a schematic. They say it's based on the JE-990 topology [Jensen 1980], which is equivalent to saying it's based on the 741 op-amp. The mundane 741 has the same topology as the 990. This topology consists of a single differential pair with active current mirror and current-source bias. The differential pair is connected to a emitter follower followed by a degenerated common-source amplifier biased by an active current source. A single-stage complementary emitter follower forms the output stage. The compensation network for the JE-990 is its most interesting feature. The dominant pole is formed in the traditional manner with a Miller compensation network on the second gain stage, but other compensation components are novel. Small inductors are placed across the degeneration resistors of the differential stage. The V_{th} (slewing threshold) of the differential stage can thus be made larger without degrading noise performance. The second-order RC network across the second-stage emitter degeneration resistor is also uncommon. Now simply scaling this up to make an ultralow-distortion power amp is not going to work. First, the output stage is going to be much more complex, with one or more stages of predrivers to drive significant power to the load. It is also hard to believe that the second gain stage in this amplifier, which swings almost from power rail to power rail, would not be a complementary design to cancel even harmonics. If it is indeed single-ended, it is almost certainly cascoded. A distortion mechanism in Miller-compensated power amplifiers [Gunderson 1984] would also have to be addressed. It is likely that some novel circuit tricks are used that the company wants to keep proprietary. That must by why the company does not want to give out a schematic.

The amplifier has an unusual system topology, with two separate closed-loop gain blocks. The first gain block has 18 dB of voltage gain. The second gain stage is said to be a scaled-up version of the first. Both stages are said to be direct-coupled, with DC servos used to reduce offsets. The manufacturer claims to have developed a complex method of safe operating area (SOA) protection. The system is said to make instantaneous power measurements of the output transistors by measuring the voltage across them and the current flowing through them. Such a system would not trip inaccurately in the presence of reactive loads. Heat sink temperature is also claimed to be considered before the amplifier is shut down.

The build quality and complexity of the 500AE reflect why this amplifier costs $3999. The chassis construction is of high quality, with a look comparable to the amplifier's competition. Thick metal is held together with machine screws. Along the two sides of the amplifier there is a staked pair of large PC boards stuffed with mil-spec or near-mil-spec components. The PC boards are high-quality double-sided boards with through holes. All the components are required because of the amplifier's two-stage design and its complex protection circuitry. The heat sinks containing the output transistors are directly beneath the PC boards on each side of the amplifier. Separate sheet metal is used to house the amplifier's large primary filter capacitors. The amp has added circuitry to prevent destructive inrush currents on power up. A single large toroidal transformer and full-wave rectifier supply power for both channels. The amplifier retails for $100 more if a multivoltage international transformer is required. No information is supplied as to the existence of any power-supply regulation in the 500AE. The inputs accept only XLR-type balanced connectors.

The most remarkable attribute of this amplifier is that the static THD-plus-noise numbers are dominated totally by noise. Into an 8-ohm load the 500AE reaches a minimum THD-plus-noise level of-100 dB at 150 watts for a 20 Hz signal. Into 4 ohms the minimum THD-plus-noise level at the onset of clipping is -100 dB at 260 watts. Dynamic distortion occurs above 10 watts, limiting at -94 dB into both 4 and 8 ohms. Dynamic distortion effects are strangely visible on both the 1 kHz and 20 kHz curves. (Remember—dynamic distortion is not a greater fault than static distortion as long as the figures are low for both, as they are here.) At clipping the 1 kHz and 20 kHz distortion is -87 dB.

The PowerCube system measured a dynamic output voltage of 37 volts (170 watts) at 8 ohms. This corresponds to the steady-state distortion of the amplifier; thus the 500AE has no dynamic headroom at 8 ohms. Only under a 1-ohm load did the voltage decline significantly. There a maximum voltage of 33 V was measured at -60° phase. A minimum of 26 V was measured at +30° phase. Given the limited information we have on this amplifier, the cause of this variation remains a mystery. Boulder literature makes a big deal of the amplifier's large phase margin, but the above result makes us wonder. Peak current output was 135 amps. Damping factor is reduced significantly at higher frequencies because of the presence of a series inductor, which improves stability at high frequencies.

So, what we have here is an amplifier that produces lower distortion than any other commercially available amplifier known to me. But achieving this performance requires very complex circuitry and hence high cost, namely $3999. In addition, the amplifier can put out only 170 watts into 8 ohms. It does behave like an almost perfect voltage source; thus it will produce very large amounts of power into low-impedance loads. By comparison, the Rotel RB-990BX (see below) is roughly one-fourth the price and 50% more powerful, but it produces ten times the dynamic distortion of the Boulder and is not as well built. (At -70.5 dB the Rotel's worst-case distortion still is vanishingly small.) If you have a Mercedes in your driveway, you can justify a Boulder 500AE in your living room, otherwise the Rotel will do just fine. On the other hand, if you have a perfectly restored Ford Model T in your garage, you might want to try a megabuck tube amp. The relationship between the performance of the antique Ford and a Mercedes is similar to relationship between a $10,000 tube amplifier and the Boulder 500AE.

Bryston 4B NRB

Bryston Ltd., 57 Westmore Drive, Rexdale, Ont., Canada M9V 3Y6. Model 4B NRB stereo power amplifier, $2195.00. Tested sample on loan from manufacturer.

This is another example of the excellence of Bryston engineering. Construction is of the same high standard we found in the Bryston preamps. In the power amplifier, double-sided PC boards with through holes are used. Only the pot used to adjust the bias current looks to be of substandard quality. When I pointed the pot out to Chris Russell, the amp's designer, he responded that it worked well in the amplifier and had never failed. It is hard to argue with well-reasoned facts. As one would expect at this price point, the amplifier is built with thick sheet metal held together with machine screws.

The 4B NRB is dual mono past the line cord. Separate power transformers drive 35A bridge rectifiers, which then drive 15,000 µF of filter caps per supply rail. Electronics for balanced-to-single-ended conversion and amplifier bridging option (for mono operation) consist of the same discrete op-amp as used in the Bryston preamp. Many of Bryston's competitors use cheap op-amps for these functions. The input signal is capacitively coupled with a film capacitor. An electrolytic cap is used in the ground return path of the main feedback loop (capacitor C_2 in Figure 6).

The amplifier's input stage consists of a pair of complementary differential pairs without any degeneration. The second stage is a complementary common-source amplifier with emitter degeneration. As in the Bryston preamplifier, the differential pairs are biased with resistors, not current sources. More surprisingly, there are no circuit additions to reduce the effect of the nonlinear base-emitter junction capacitance of the second gain stage. Normally an emitter follower buffers the first gain stage from the second, and/or the second gain stage is cascoded. This is required because the voltage swings at the collector-base junction of the second stage of a power amplifier are rather large, coming close to the power supply rails of the amplifier when it is delivering full power. This results in significant variation of the collector-base junction capacitance. If the second stage is not buffered from the first, the nonlinear capacitance will result in a change in the amplifier's high-frequency open-loop gain as the output of the second stage moves across its voltage range. This leads to dynamic distortion [Cherry 1982], [Cordell 1980], [Borbely 1989]. The disadvantage of the follower stage is that it introduces another open-loop time constant, which can degrade the amplifier's closed-loop performance. Bryston was able to eliminate the need for the emitter follower by using the same technique as in their preamplifier. Second-order distortion is reduced by minor circuit changes, such as unconventional bias currents and simple passive circuit additions.

Supply voltage regulation is in the amplifier's first two stages to increase power-supply rejection ratio and desensitize these stages to supply-rail variations that occur as the amplifier is required to source or sink large amounts of current. A zener-diode shunt regulator is employed. These circuits can work on a lower-voltage supply rail than the output stage, without reducing the amplifier's output swing, because the final stage of this amplifier has a gain of 3. The required voltage swing of the second stage is thus reduced by a factor of 3 and it remains well within the regulated supply rails.

Adding gain to the output stage of a power amplifier is a nontrivial matter because of stability issues which occur when an amplifier has more than two voltage-gain stages. The amplifier can also become more sensitive to reactive loads. So, while Bryston's competitors were adding Wonder Caps to tube circuits developed during World War II, Chris Russell and his staff had to create a new output topology for the 4B. To keep the output circuit stable and set to its desired gain, nested feedback was chosen for the 4B. In addition to supplying voltage gain, another goal of the Bryston output stage was to reduce distortion in the crossover region resulting from mismatches between the complementary *pnp* and *npn* devices. The complete output topology used by Bryston is very clever and quite complex. The amplifier's output terminal is connected to two different transistor stages in parallel. One stage is a common-emitter topology, the other a common-source topology. Three stages of predrivers precede the final output stage electronics to reduce loading on the second gain stage. Bryston has a nice two-page technical description of the output stage, including schematic. To save space here, I direct interested readers to the Bryston publication. (Now, if other manufactures would produce literature as detailed as Bryston....) The actual circuit has an additional predriver stage, which is not shown in the simplified schematic. A rather complex current limiter is used for the output stage, but this is not shown in the Bryston document. In addition, the output stage is cascoded in the 4B to reduce the V_{CE} of the transistors by a factor of 2. This is accomplished by applying to the base of the cascodes a voltage which is proportional to the amplifier's output signal. The smaller Model 3B has lower power-supply rails and does not use the cascode stage. A total of 28 transistors per channel is used to implement the power amplifier, with an additional 16 devices used for the balanced input and bridging circuits.

Compensation is principally accomplished by creating a dominant pole at the second gain stage with a single capacitor to ground. The complete compensation network topology shows more Bryston innovation, but its operation is proprietary, so I cannot discuss it here. A 2 µH inductor is in series with the output to prevent the amplifier from going unstable when loaded by excessive capacitance. Damping factor at high frequencies is reduced by the presence of this inductor. A relay clamps the output of the second stage to ground on power up. The relay releases when the amplifier is stable. A separate circuit, consisting of a special-purpose integrated circuit, an op-amp, an SCR, and an optocoupler, slowly ramps up the line voltage to the amplifier on power up. This ensures that components will not be damaged by inrush currents. The need for this circuit becomes clear when you realize this amp is warranted for 20 years. The approach is more sophisticated than switching in a resistor in series with the power line, as done in the more expensive Boulder 500AE. Thermal breakers are in series with the Bryston's power line for protection against excessive temperature. The only fuses are external line fuses.

Into an 8-ohm load the Bryston 4B NRB reaches a minimum THD-plus-noise level of -96.5 dB with a 1 kHz input and -94 dB with a 20 Hz input, both at 250 watts. At this point the amplifier clips, so this measurement is dominated by noise. The Boulder 500AE achieves an even lower minimum level because it has a lower noise and hum level. Into 4 ohms the minimum THD-plus-noise level of the Bryston at the onset of clipping (400 watts) is -88 dB at 1 kHz and -83 dB at 20 Hz. Above 5 watts dynamic distortion becomes evident, with the 20 kHz distortion curve flattening out and reaching a minimum of -86 dB with an 8-ohm load. The 20 kHz distortion then rises to a preclipping level of -83 dB. Into a 4-ohm load the 20 kHz distortion curve is again just about flat between 5 watts and clipping, reaching a minimum of -81 dB. While the overall distortion performance is good, and is better than specified by Bryston, it is somewhat disappointing given the technical sophistication of the circuit. The PowerCube system measured a dynamic output voltage of 52 V (338 watts) at 8 ohms. This corresponds to 1.3 dB of dynamic headroom. The PowerCube showed that the maximum voltage output of the amplifier declined by 34% into 2 ohms and 56% into 1 ohm. The current limiter was reasonably well-behaved, since output varied 3% to 17% across the five phase angles in each of the four tests. Peak current output was 52 amps.

As will be seen from the Rotel review below, a power amp with performance similar to that of the Bryston 4B NRB can be had for half the price. What you get for twice the price is build quality that allows the unit to be warranted for 20 years. More important, you get a design which is bulletproof enough to withstand the rigors of professional sound-reinforcement work. In such an application, down units are simply not acceptable. As an example of this, we accidentally drove the amp into clipping with a 100 kHz signal while it was connected to a 4-ohm load. After replacement of the external power fuses, the amp worked perfectly. If a dead amplifier is not acceptable to you, you should seriously consider this unit.

Dynaco Stereo 400 Series II

Dynaco, a division of Panor Corporation, 125 Cabot Court, Hauppauge, NY 11788. Stereo 400 Series II power amplifier, $995.00. Tested sample on loan from manufacturer.

Those of you over 30 no doubt remember the name

Dynaco. The company produced low-cost kits that represented good value at the time. Dynaco products were marketed as close-to-the-best-for-less. Most owners of Dynaco products probably remember them less than fondly. It did not take long for a Dynaco to develop noisy switches and controls, and for the gold-colored face plate to tarnish. On the other hand, this most successful marketer of mass-market kits was no doubt responsible for kindling an interest in electronics in a great many teenagers. The old Dynaco company is long out of business. With the advent of modern construction techniques, it costs more to produce a kit than a built unit. Even Heathkit, which held on for several years after Dynaco, has recently stopped producing kits. Why then Panor Corporation has decided to revive the company's name as a manufacturer of relatively expensive audio components is unclear to me. Maybe they figured that the name still had some recognition in tube equipment, which is what they came out with first. The Stereo 400 Series II is their first new solid-state product. My evaluation of the product is of course independent of its name.

This Dynaco amplifier is a conventional design. Complementary differential pairs biased by current sources form the first stage. The differential pairs are degenerated, but the degeneration resistors are bypassed with capacitors. This method of introducing a transmission zero into the forward path as part of the compensation network is strange, since the nonlinearity of the first stage at high frequencies is increased. The same method of widebanding a differential stage was used by Otala in one his early designs [Lohstroh and Otala 1973], but it is unclear if he would still recommend it. (Note that Boulder does just the opposite, placing an inductor across the degeneration resistor.) The second stage is a complementary common-source amplifier with emitter degeneration. As in the Bryston, there are no circuit additions to reduce the effect of the nonlinear base-emitter junction capacitance of the second gain stage. No novel circuit tricks to reduce this effect can be seen in the schematic. The second stage is followed by a complementary source-follower predriver. The final output stage consists of a complementary source follower, the active stage being realized with four bipolar devices in parallel. The output stage is biased by a two-transistor V_{BE} multiplier. The output of this stage is connected to the speaker terminal through a series inductor. High-frequency stability is improved at the cost of some reduction in high-frequency damping factor. Current limiting occurs on the second voltage-gain stage. The principal action of the circuit would appear to be to prevent damage to this gain stage when the latter is near clipping. Total transistor count per channel, including paralleled devices, is 28. Compensation is principally accomplished by creating a dominant pole at the second gain stage with a pair of capacitors to ground at each side of the V_{BE} multiplier. A 470 µF electrolytic cap is used in the ground-return path of the main feedback loop (capacitor C_2 in Figure 6). Another 22 µF electrolytic capacitor is used at the input (C_1). Both electrolytics are bypassed with film capacitors. The output stage of the amplifier is protected by supply-rail fuses. The bias current to the differential pairs is removed on thermal overload.

The death of the electronic kit is clearly understandable when you see how this unit is put together, using modern construction technology. Almost everything is mounted to the PC board, and very little point-to-point wiring is identifiable. The little handwork that is required is done with the highest professionalism one expects in onshore assembly. Solder joints are well-flowed, and shrink-wraps are used on all coax cables. Components on the PC board are mostly machine-inserted and soldered. The PC board is double-sided with through holes. A single toroidal transformer is shared by both channels. One diode bridge is shared by both channels. The primary filter caps are 24,000 µF. The transformer and heat sinks are significantly smaller than on the similarly priced Rotel. Predrivers on the PC board also have very small heat sinks. A front-panel switch reconnects the secondary winding of the transformer. This is called the high-current mode. The transformer's output voltage is lowered, but its series losses are also reduced in this mode. More power may be available to a low-impedance load in the high-current mode. The overall build quality of the unit is not up to the standards of the costlier Boulder and Bryston. The cabinet is made of relatively thin metal. It is held together with cheap sheet-metal screws. We were unable to reseat some of the top-plate screws after they had been removed. Component quality is not always mil-spec, but no component appears to be underspecified.

Into an 8 ohm load the ST400II reached a minimum THD-plus-noise level of -86 dB at the clipping point of 205 watts with a 1 kHz input. The 20 kHz distortion deviated from the 1 kHz curve even at 10 mW. The minimum level of 20 kHz distortion just before clipping was no better than -68 dB. It is not clear why this result was substandard; perhaps the problem is due to the compensation components discussed above. The 4-ohm measurements were made with the transformer in the high-current mode. Into 4 ohms the minimum distortion level at the onset of clipping was -82 dB. The left channel clipped cleanly at 180 watts. The right channel had a softer clipping characteristic, starting at 120 watts. The 20 kHz numbers were again much higher at all power levels. The minimum level before clipping was -62 dB. The PowerCube measurements were always better in the normal-current mode and thus are the ones reported here. A dynamic output voltage of 54 volts (364 watts) at 8 ohms was measured. This represents a dynamic power headroom of 2.5 dB. The voltage declined 14% into 4 ohms, 57% into 2 ohms, and 79% into 1 ohm. Into a resistive load, the dynamic power at 1 ohm was 126 watts. Peak current output was 17 amps in both transformer positions.

The dynamic power measurements of the ST400II appear to show that the design is underbuilt. Construction quality of the unit appears to confirm this. The 20 kHz distortion figures are also high throughout the power band. For $105 dollars more the Rotel unit (see below) handily outperformed the Dynaco—in some cases by an order of magnitude! We thus cannot recommend this unit.

Hafler Series 9500 Transnova

Hafler, a division of Rockford Corporation, 648 River Street, Tempe, AZ 85281. Series 9500 stereo power amplifier, $1800.00. Tested sample on loan from manufacturer.

This is the only power amp in this survey to use a MOSFET output stage. Indeed, this is the only power amp in the survey to use FETs of any kind. (In truth, however, I do not know what devices are used by Mr. Yee in the UltrAmp). The Hafler marketing department has been pushing the MOSFETs-are-like-tubes angle in their advertising. This sullies the fine work of Jim Strickland, who is following in the path of Erno Borbely. Borbely was an early employee of Hafler (long before it was taken over by Rockford) and had a preference for MOSFETs for good scientific reasons (see above).

As stated above, the distortion characteristics of a MOSFET used as a source-follower output stage will be higher than those of a comparable bipolar stage. One way around this is to use the MOSFET in a common-source configuration. Headroom is also significantly improved with the common-source configuration. However, there are several problems with that approach. The most significant of these is the problem of biasing the devices. Traditional V_{BE} multiplier configurations cannot be used because the biasing loop now includes the power supplies. Any variation in the power supply now affects the output quiescent current. This problem has been attacked numerous times in CMOS integrated circuits. The solution usually involves the addition of a complete error amplifier around each output transistor. The error amp sets the quiescent current of the transistor and defines its gain. Assuming that each error amp sets the same current in both sides of the complementary pair (additional circuitry is often required to ensure this assumption is valid), the quiescent current of the output is set. Jim Strickland's solution to this problem is much simpler but far less obvious. His innovation is a dynamic power supply.

Here is how it works. Both sources of the complementary MOSFETs are connected to ground. The gates of the MOSFETs can now be biased and driven by the second stage in the standard manner. A stacked diode array in series with the collectors of the second-stage transistors sets the fixed voltage difference across the gate transistors, thus establishing the quiescent current. Now the drain of the *p*-channel output-stage MOSFET is connected to the positive supply rail and the drain of the *n*-channel MOSFET is connected to the negative supply rail. Think about this for a second—what happened to the positive output terminal of the amplifier? It's at the center tap of the power transformer! What happens in the amplifier is that as the output moves the transformer secondary, the full-wave rectifier moves and the power-supply filter capacitors move. The whole power supply follows the output signal! Clearly this amplifier cannot be described as direct-coupled.

With the bias problem solved, the next issue to deal with in a common-drain amplifier is the output stage's voltage gain. With three stages of voltage gain, there will be three high-frequency poles and lots of open-loop gain [Grebene 1984]. This is a recipe for an oscillator, not an amplifier. The solution used in the Hafler is to use a nested feedback loop in the output stage [Grebene 1984]. This loop is formed by a resistor between the output terminal and the gate of the output stage. This local shunt feedback loop stabilizes the output stage's transresistance [Gray and Meyer 1984]. Another problem with a common-drain output stage is that it has a very high output impedance [Grebene 1984]. This makes the gain of the stage and its high-frequency transfer characteristic highly dependent on the value of the amplifier's load. The aforementioned shunt feedback loop lowers the effective output impedance of the stage to help reduce this problem. The amplifier's name, Transnova, apparently is a reference to the output stage's transresistance property. The shunt feedback also reduces the input impedance of the output stage. Since the second gain stage has a high-impedance output, the voltage swing at the input to the output stage is limited. This allows the first and second stage to run on 24 V regulated rails. Power MOSFETs have large values of gate-to-source capacitance. When wired in a source-follower configuration this capacitance is bootstrapped, lowering its effective value. In the common-source configuration the capacitance is Miller-multiplied. This creates a difficult load for the second gain stage to drive.*

*Much of my statements (with supporting references) in this paragraph will be found to be in contradiction to a paper by Cherry [Cherry and Cambrell 1982]. In his paper a formal analysis of both common-emitter and emitter-follower amplifiers is undertaken. From mis mathematical analysis Cherry concludes that the stability of an amplifier with a common-emitter output stage should be very similar to that of an amplifier with an emitter-follower output stage when a load is attached. Other amplifier characteristics, including output resistance and distortion, are also claimed to be similar. This runs counter to my experiences, and I believe that Cherry's analysis may be flawed because of the simplification required to produce usable analytical results.

For example, the emitter-follower amplifier model used by Cherry does not include any predriver stages, causing a significant fraction of the load impedance to be reflected to the second gain stage. Another example is that the parasitic capacitor across the output device in the common-emitter amplifier is analyzed by replacing it with a Miller-multiplied capacitor at the input to the third stage. This capacitor also gives rise to a right half-plane zero [Gray and Meyer 1984], a further source of stability problems, but this zero is not considered in the Cherry paper. Cherry also claims that nested feedback around the third gain stage does not improve stability or enhance performance.

Professor Cherry is one of the seminal thinkers in audio, so he is not very likely to be wrong. I would therefore welcome any of our technical readers to comment on his paper.

The front end of the Series 9500 is more conventional than its output stage. Complementary differential pairs with JFETs form the first stage. The *n*-channel sources are connected to the *p*-channel sources through a resistor. Because the JFETs have a negative threshold (for the *n*-channel device), this arrangement self-biases the differential pairs. Unlike a constant-current biasing scheme, the current in the differential pair can increase when the differential pair is driven with a large differential current. This improves large-signal dynamic performance. A similar circuit was developed by Sansui [Takahashi and Tanaka 1984] using bipolar devices but is much more complex because the bipolar will not self-bias. The second stage is a complementary common-emitter stage. Both the first and second stage are cascoded with bipolar devices. There is a total of 19 transistors, including the quadruple paralleled output devices. Feedback is taken from the positive terminal of the amplifier (the transformer's center tap) back to the non-inverting differential-pair input, using the standard passive resistor divider. A 220 µF electrolytic cap is used in the ground return path of the main feedback loop (capacitor C_2 in Figure 6). It is bypassed with a small film cap. The amplifier's dominant pole is formed by a Miller capacitor around the second gain stage. Additional secondary compensation circuits are used throughout the amplifier. They are required keep the three-gain-stage topology stable.

A single huge transformer is used in the Series 9500. Each channel has its own secondary, which is connected in the dynamic configuration described above. The supplies are filtered with 20,000 µF capacitors, each paralleled with a 4.7 µF film capacitor. The first and second stages are driven by ±24 V regulated power supplies. The low power-supply rails can be used because the output stage has voltage gain. The regulated supply starts with a button-sized full-wave rectifier. 1000 µF capacitors filter the rectifier's output. LM317 and LM337 integrated rectifiers are used to generate the regulated rails. This supply is shared by both channels. I was somewhat surprised that separate regulators were not used for each channel. Owing to the more robust nature of MOSFETs, the only protection devices on the amplifier are the power-line fuse and fuses in the dynamic supply rails. A turn-on delay circuit prevents current flow in the differential pairs until the output stage has settled.

Construction quality of this amplifier is excellent. Thick sheet metal is held together with high-quality machine screws. Custom-designed heat sinks start at the side of the amplifier and then curve to the back. Double-sided circuit boards are stuffed with high-quality components. A large metal bar is placed across the inboard side of the output devices to ensure good mechanical contact with the heat sinks.

Given all the innovations in this amplifier, I was hoping to see static distortion numbers that would rival those of the Boulder 500AE, Bryston 4B, and Bob Cordell's MOSFET design. It turned out this was not going to occur. Into an 8-ohm load the 990 reaches a minimum THD-plus-noise level of -81 dB at 240 watts with a 1 kHz input. Into 4 ohms the minimum distortion level at the onset of clipping (400 watts) is -77 dB. At 0.55 watts the 20 kHz distortion curve reaches a minimum of -70 dB, then rises and plateaus, reaching -60 dB at clipping into 8 ohms. Into a 4-ohm load the 20 kHz distortion reaches a minimum level of -67 dB and rises to -57 dB at clipping. The origin of the relatively high 20 kHz distortion is unclear. It may arise from the dynamic power supply, or it could be an indication that the second stage is having trouble driving the capacitive load of the output stage. The 10 kHz square-wave response into a 6-ohm load with a capacitive component of -45° showed more than average overshoot and lots of ringing. Capacitive-load square-wave testing of an amplifier with an inductor in the output stage is not revealing of amplifier instability because the LC resonance dominates the amplifier's response. This amplifier does not have an inductor in the output stage, but the transformer is of course inductive. The value of this inductance and the value of the large bypass capacitors would not cause ringing at the rate observed in this test. As stated above, stability problems are more likely in a three-stage design.

The PowerCube system measured a dynamic output voltage of 55 V (378 watts) at 8 ohms. That represents 1.8 dB of dynamic headroom. The PowerCube showed that the maximum voltage output of the amplifier declined by 20% into 2 ohms for noninductive loads and by 45% into 1 ohm. The dynamic power into a 1-ohm resistive load measured 894 watts. Available output voltage increased into reactive loads; thus no stability problems were identified in the PowerCube tests. No *I-V* current limiter artifacts were observed because the Series 9500, like most MOSFET amps, does not require an *I-V* current limiter. Peak current output was 71 amps.

I wanted this amplifier to be recommendable, given all the innovative circuitry and good build quality for the money. The 20 kHz distortion and some evidence of lower stability margins into capacitive loads militate against such a recommendation. For only 1.7 dB more cash ($395, that is), the Bryston gives you 20 dB less 20 kHz distortion, the same level of construction, the same maximum power, and balanced inputs. I hope Jim Strickland can overcome the remaining design problems of his topology in the next generation of this product. In the meantime, does anybody out there want to produce a scaled-up version of Bob Cordell's state-of-the-art MOSFET power amplifier?

R.E. Designs LNPA 150

R.E. Designs, 43 Maple Avenue, Swampscott, MA 01907. LNPA 150 monoblock power amplifier, $2700.00 the pair. Tested sam-

pies on loan from manufacturer.

This is a new and very small company. The product is advertised in the classified advertising pages of *Audio* magazine. The LNPA 150 is a monoblock design, differing from other amps in this survey and almost all power amps in general in that it has all electronics, including the output stage, on regulated rails. This is a very expensive undertaking, since what we have here is in essence one power amplifier (the regulated power supply) driving another power amp. The design of the regulated amplifier is somewhat simplified, since it needs only to source (sink) the positive (negative) supply voltage. The maximum power that must be dissipated by the regulator is limited to the product of two quantities: maximum current to be delivered to the load times the difference between the unregulated and regulated supply rails. The output stage devices may be required to dissipate power which is equal to the load current times twice the regulated rail voltage. The advantage of a regulated supply is that the amplifier is independent of supply-line variations and power-supply noise. The power-supply rail does not change value under changes in load conditions. The disadvantage, beyond significantly increased cost, is the loss of any dynamic headroom. Except for independence from power-line variations, these advantages can be gained by regulating only the voltage-gain stages of the amplifier. Such regulators (used by Hafler and Bryston) are much simpler to design because they are not required to source significant current. Didden has shown that dynamically varying the output-stage power supply can actually be advantageous, since the maximum V_{CE} on the devices is limited and higher-speed output devices can be used [Didden 1983].

Separate transformers and bridge rectifiers are used for the positive and negative supply rails of the amplifier. The unregulated filter capacitor is 33,000 µF. The voltage references for the supply regulators are LM317/337 devices. The regulator is formed with an LM343 op-amp and a discrete triple emitter-follower output stage. 100 µF capacitors are used on the regulated rails.

The actual amplifier is mostly conventional. Essentially, the much less expensive Rotel (see below) uses a scaled-up version of the same topology—and produces a lot more power. Complementary differential pairs are biased with a resistor and are not degenerated. The output of the differential pairs is then connected directly (no emitter follower is used as a buffer) to a complementary common-emitter stage with emitter degeneration. The principal novelty of the design is the termination of the collector of the differential-pair transistor—the one that has its base connected to the inverting input of the amplifier—into the emitter of the second gain stage. This enhances the open-loop voltage gain of the amplifier. The second gain stage then drives the output stage, which consists of a pair of complementary emitter followers driven by a single complementary emitter-follower predriver. A V_{BE} multiplier is used to bias the output stage. A dominant pole for frequency-compensating the amplifier is created by connecting a capacitor from the output of the second gain stage back to the inverting input. Bipolar devices are used exclusively in this design. The active amplifier uses a total of 14 devices. An electrolytic cap is used in the ground-return path of the main feedback loop (capacitor C_2 in Figure 6), but no other caps are in the signal path.

Construction quality leaves a lot to be desired, given the amplifier's price. This is in no way due to price gouging by R.E. Designs but is simply a consequence of their very small-volume production. The result is an amplifier that looks a lot more like an *Audio Amateur* construction project than a Krell. Most surprising was that the terminals of the power transistors were accessible through the bottom of the heat sink. This is a safety hazard that the manufacturer claims will be corrected in future production runs. PC boards are single-sided and hand-soldered. They are stuffed with high-quality components.

Distortion performance of the amplifier proved to be below average. Into an 8-ohm load the unit reaches a minimum THD-plus-noise level of -85 dB at 0.15 watts with a 1 kHz input. This then rises gently to -69 dB at the onset of clipping with 70 watts output, full clipping being at 85 watts. Into a 4-ohm load the THD-plus-noise level at the onset of clipping (140 watts) was -66 dB. Distortion levels at 20 kHz were significantly higher. The 20 kHz distortion was never lower than -79 dB into 8 ohms and -72 dB into 4 ohms. At clipping the 20 kHz distortion was -51 dB into 4 and 8 ohms. The manufacturer argues that these values are still below the level of audibility. *[All 20 kHz distortion is inaudible, regardless of magnitude, because the second harmonic is at 40 kHz, the third at 60 kHz, etc.; but serious 20 kHz distortion is indicative of other problems.—Ed.]* According to the manufacturer the distortion performance was traded for improved signal-to-noise performance in the design of this amplifier. Lowering noise levels in an amplifier often involves basic design changes—such as changing the bias levels of devices—that can degrade distortion performance. The R.E. Designs amplifier did exhibit the lowest noise level of the group here, but the Boulder amplifier had a noise floor only 8.5 dB higher.

The lower noise levels are claimed by the manufacturer to make an audible difference. We ran a series of ABX comparison tests of the R.E. Designs amplifier against the Rotel amplifier (see review below). Three experienced listeners, in sessions lasting about one hour per person, obtained totally random results. The units could not be distinguished.

The PowerCube system measured a dynamic output voltage of 26.7 V (89 watts) into 8 ohms. This is the same as the steady-state value for the amplifier at 1% distortion (the PowerCube's target level). This is the expected result for a fully regulated amplifier. The PowerCube showed that the maximum dynamic voltage output of the amplifier declined by 11% into a 2-ohm resistive load and 44% into 1 ohm. Slightly higher dynamic power is available into reactive loads. Peak current output was 28 amps.

When, after evaluating this amplifier, we realized it was not going to receive a particularly good review, we contacted the manufacturer to see if we could just return the pair of samples unreviewed. As stated previously, *The Audio Critic* does not want to harm a micro manufacturer with a bad review. Dan Banquer, the company's president, stated that he wanted the review to run regardless of its outcome. The outcome is that the amplifier cannot be recommended. Its construction quality and power output are not commensurate with its price. The amplifier's very low-noise output is an important achievement of the design, but this alone cannot overcome the other negatives discussed above.

Rotel RB-990BX

Rotel of America, P.O. Box 8, North Reading, MA 01864-0008. RB-990BX stereo power amplifier, $1100.00. Tested sample on loan from manufacturer.

This unit turned out to be the big surprise in this survey. In most respects it performed almost as well as the state-of-the-art designs by Boulder and Bryston, but it costs only $1100. Unlike those amps, the Rotel is a conventional design. Complementary differential pairs biased by current sources form the first stage. The differential pairs are not degenerated. The second stage is a complementary common-source amplifier with emitter degeneration. As in the Bryston, there are no circuit additions to reduce the effect of the nonlinear base-emitter junction capacitance of the second gain stage. No novel circuit tricks to reduce this effect can be seen in the schematic. The second stage is followed by a pair of complementary source-follower predrivers. The final output stage consists of complementary source followers, with the active stage realized by five bipolar devices in parallel. The triple Darlington output stage offers excellent isolation of the second gain stage from the load and does not have the stability problems associated with output stages that have local feedback loops. The disadvantage of the stage is that it has higher distortion than other triple output stages [Bongiorno 1984]. The output of this stage is connected directly to the amplifier output terminal without a series inductor. This keeps the high-frequency damping factor from declining but at the risk of reduced amplifier stability into high-frequency loads. The five paralleled transistors on each side of the output stage are biased by a single-transistor V_{BE} multiplier. No *I-V* current limiting is used. Total transistor count per channel, including paralleled devices, is 23. Compensation is principally accomplished by creating a dominant pole at the second gain stage with a single capacitor to ground. A 100 µF electrolytic capacitor is used in the ground-return path of the main feedback loop (capacitor C_2 in Figure 6). Another 50 µF electrolytic capacitor is used at the input (C_1).

The amplifier is protected by supply-rail fuses. During one of our large-signal tests the fuses were blown in one channel. After they were replaced, the amp was fully functional. The fuses alone thus appear adequate to protect the amplifier. The downside of this approach is that the unit must be physically opened to replace the fuses after a fault condition. There is also the danger that both fuses will not blow simultaneously. If this condition occurred, the result could be significant damage to the amplifier. The input of the amplifier is shorted by a relay if the heat sinks go above a preset temperature limit. The relay is also closed on device power-up.

A single, very large, shielded, toroidal transformer is shared by both channels. Separate full-wave rectifiers and filter caps are used for each channel. Each supply rail has 15,000 µF of capacitance across it. Construction quality of the unit is not up to the standards of the much costlier Boulder and Bryston. The cabinet is made of relatively thin metal. It is held together with cheap sheet-metal screws. Component quality is not always mil-spec, but no component appears to be underspecified. Single-sided PC boards are the most glaring but not the only sign of mass-market assembly techniques. Copper bus bars can be seen on the PC board; they are a kludge to reduce trace resistance.

From the above circuit analysis we expected the amplifier to exhibit more distortion than a more complex topology, such as that of the Boulder 500AE or the Bryston 4B. The results below show that this turned out to be the case but just barely; the distortion levels are still very low. Into an 8-ohm load the RB-990BX reaches a minimum THD-plus-noise level of -82 dB at 200 watts with a 1 kHz input. Into 4 ohms the minimum THD-plus-noise level at the onset of clipping (310 watts) is also -82 dB. The distortion curves are dominated by noise below 100 watts. Above 10 watts the 20 kHz distortion curve flattens out at a level of -74 dB and stays there until clipping into 8 ohms. Into a 4-ohm load the 20 kHz distortion curve flattens out at 10 watts at a level of -70.5 dB and stays at that level until clipping. The PowerCube system measured a dynamic output voltage of 47 V (276 watts) into 8 ohms. This closely corresponds to the steady-state power of the amplifier at 1% distortion; thus the amplifier has very little dynamic headroom at 8 ohms. The PowerCube showed that the maximum voltage output of the amplifier declined by only 12.5% into 2 ohms and by 25% into 1-ohm noninductive loads. The dynamic power into a 1-ohm resistive load measured 1220 watts. Driving heavily inductive loads did have some effect on power output. Driving a 1-ohm load with a +60° phase angle resulted in a 38% loss of voltage relative to a resistive load. The reason for this behavior is unclear to me, although I am told it has been seen before by the designers of The PowerCube. The amplifier apparently does not like such a highly inductive load. The result is increased amplifier distortion or oscillations, causing the PowerCube to stop the test. Even though the amplifier has no inductor in series with its output, the 10 kHz continuous-time square-wave response into a 6-ohm load with a capacitive component of -45° was reassuring, the amplifier showing good stability into this load. Peak current output was 211 amps, almost twice the value of any other amplifier in this survey!

To use the term coined by *Consumer Reports,* this is a Best Buy. It puts out huge amounts of power with little distortion. To achieve the $1100 price, some things had to be compromised. Construction quality is more than adequate but it is not at the same level as in the Boulder or Bryston. The circuitry is less complex. Protection circuits are not as sophisticated and balanced line inputs are not included. For the vast majority these compromises will be entirely acceptable.

UltrAmp Power Amplifier

Mobile Fidelity Sound Lab, 105 Morris Street, Sebastopol, CA 95472. UltrAmp Power Amplifier (stereo), $1295.00. Tested samples on loan from manufacturer.

This is the third of a trio of components developed by Michael Yee for Mobile Fidelity Sound Lab and sold directly to the end user. We found the first two of these products to be poor performers (see Issues No. 18 and No. 19). As we shall see, this power amp is no exception. In the meantime Mobile Fidelity has discontinued the line; if you call the UltrAmp "800" number, they tell you that the line is being reengineered but that the old units are still available on special order. Mr. Yee is now selling electronics under his own name, Michael Yee Audio. Except for a change in silk-screening, the Michael Yee Audio products look identical to the products originally sold by Mobile Fidelity. I have no detailed technical information on the insides of these units, old or new, other than the companies' product literature. New Michael Yee Audio literature continues to claim major sound-quality improvement for these products, based on breakthrough design techniques. As will be seen from the measurements, this amp may indeed sound different but not because of a major breakthrough.

Measured results are consistent with other products by Michael Yee—not good. Into an 8-ohm load the amp reaches a minimum THD-plus-noise level of -77 dB at 50 watts with a 1 kHz input, then clips at 90 watts with -74 dB distortion. Above 1 watt the 20 kHz distortion curve flattens out at a level of -60 dB and stays there until clipping, where the distortion becomes -52 dB. Into a 4-ohm load the THD-plus-noise level with a 1 kHz input reaches a minimum of -72 dB at 30 watts, then rises to -70 dB at 63 watts, where soft clipping begins. (Note that this amp produces less power into 4 ohms than 8 ohms!) At a -40 dB distortion level the amplifier puts out 100 watts into 4 ohms. The 20 kHz distortion into 4 ohms remains essentially flat at a -54 dB level from 0.4 to 70 watts. The PowerCube system measured a dynamic output voltage of 30 V (112 watts) at 8 ohms. This closely corresponds to the steady-state power of the amplifier at 1% distortion; thus the amplifier has very little dynamic headroom at 8 ohms. At 4 ohms the PowerCube measured a 30 V output into a +30° inductive load, but only 20 V into a resistive or a -30° capacitive load. The output dropped to 15 V with a ±60° load at 4 ohms. Into a 2-ohm load the output voltage had a similar characteristic, ranging from 6.6 to 12.5 V, depending on the reactive component. Into 1 ohm the output ranged from 3 to 4.1 V. With a 1-ohm resistive load this amp could supply only 15 watts. Now here is an amplifier which can claim to perform like a tube amp! Clearly there is a major problem with the current limiter in this design. Connect this amplifier to a loudspeaker with a highly complex load impedance, and it will sound different from the others in this survey. Peak dynamic current output measured a scrawny 6 amps.

Readers will recall that the first sample of this amplifier blew up immediately upon power-up (see Issue No. 18, page 38). The second sample of the amplifier came to exactly the same destructive end, shortly after our measurements had been completed. This happened while the unit was idling—powered up but with no source active. Big bang, huge puff of smelly smoke, dive for the power cord! Mobile Fidelity Sound Lab was apparently smart enough to cut their losses with Michael Yee. You can avoid any losses at all by not purchasing a Michael Yee Audio product, at least not until Mr. Yee gets his act together.

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Three Speaker Systems in the 2.5 to 3.5 Kilobuck Range: Two Big and One Small

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While the tweako cultists are dreaming about, and saving for, the latest high-priced tube amplifier, the real designers of the audio world are coming up with loudspeakers that actually sound better.

My introductory comments preceding the loudspeaker reviews in the last issue (No. 19, p. 11) are still fully applicable here; I see no reason to repeat them. If you are new to ***The Audio Critic,*** I recommend that you acquire all issues beginning with No. 16 in order to become conversant, and comfortable, with our reviewing philosophy.

I wish to add only one thing I may not have made entirely clear before. Loudspeakers differ so widely in frequency response and wave-launch geometry that they are readily distinguishable without the ABX listening comparisons we consider so essential in our evaluations of purely electronic equipment. Not that ABX-ing loudspeakers (i.e., comparing them double-blind at matched levels) is without value—far from it, witness the highly effective test procedures at Canada's NRC facility. There are formidable problems to overcome, however, when setting up such tests—concealment of visual/positional/directional clues, level matching of nonflat outputs, etc.—and so far we have done only a few tentative experiments. Two reasonably flat speakers of approximately the same size and configuration can sometimes sound disturbingly similar at matched levels, but the speakers reviewed below are so different from their competition in design and measurable output that valid sonic evaluations of them can easily be made without ABX-ing.

—Ed.

DCM TimeWindow Seven

(Reviewed by Peter Aczel)

DCM Corporation, 670 Airport Boulevard, Ann Arbor, MI 48108. TimeWindow Seven floor-standing 3-way loudspeaker system, $2999.00 the pair. Tested samples on loan from manufacturer.

In 1977, this magazine in its earlier incarnation "discovered" and enthusiastically recommended the original DCM TimeWindow; to my knowledge, no other publication at the time was aware of it. The rest is history: the TimeWindow became the darling of the high-end-sound-at-an-affordable-price market; that lasted a couple of years; later the company fell on hard times, then gradually bootstrapped itself to a viable position again; for the past few years they have been supplying chain outlets with well-engineered economy speakers, but even so they have always kept one avatar or another of the TimeWindow in the line as their flagship.

The Seven is by far the most elaborate and most expensive version of the TimeWindow so far. It has the signature of Steven J. Eberbach, the designer, on a plaque in the back, indicating that he considers the speaker to be representative of his best work—and it is. The "flying wedge" configuration of the original TimeWindow has been retained, with the apex of the wedge in front and pairs of drivers mounted symmetrically on the two vertical planes formed by the wedge, but there the resemblance stops. The TimeWindow Seven is much larger than the original; it is a full four feet high and weighs 70 pounds, although its footprint is still relatively small; the overall look is much more high-tech, with black lacquer top, black lacquer vertical fluting, and black grille cloth; and the price is four and a half times that of the 1977 version (which is actually not so much higher when adjusted for inflation). Most important—in terms of performance there is no comparison. This newest TimeWindow is, to paraphrase W. S. Gilbert, the very model of a modern major loudspeaker, a loudspeaker for the CD era.

The driver complement of the speaker consists of a pair of coaxial units, each incorporating a ¾" hard-dome tweeter and a 6½" polypropylene cone driver; a pair of 9" woofers; and a single rearward-firing ¾" hard-dome tweeter (à la Snell). The woofers are enclosed in what the manufacturer calls a "rear-ported hybrid chambered transmission line" and what to me looks like, and behaves like, a vented box with a fat tube tuning the vent. The passive network that routes the signal to the drivers is quite a complicated affair, since Steve Eberbach believes in compensating for the time/phase differences between electrical input and acoustical output to the greatest extent possible. That's his Thing, as you can also discern from the coaxial configuration, not to mention the design of the original Time Window. The crossover frequencies are not given; my guess is 500 Hz and 4 kHz; the rear tweeter appears to have only a series capacitor on it and also comes in at approximately 4 kHz. Three spectral balance controls in the back of the speaker permit a narrow (and therefore safe) range of adjustment of the high frequencies, midrange, and lower midrange. Factory reference settings are provided.

The left and right speakers are not identical; they are designed as mirror images of each other, the matched pairs being the inboard and outboard panels. When the apex of the wedge is aimed forward, the inboard panel is aimed 35° off the wedge axis into the listening area, and the outboard panel is aimed 35° off the wedge axis outward, toward the periphery of the room. Here comes the sophisticated part: the outboard coaxial unit is set 6 to 10 dB lower in output level than the inboard one, so that the direct sound of the inboard unit dominates the listening area and the outboard unit becomes a kind of ambience processor. The idea is almost in direct opposition to that of a monitor speaker (a designation usually applied to simple, forward-firing, single-sound-field, simon-pure designs), but the results are excellent—the speaker produces a big, room-filling sound with lots of soundstage detail and without any audible beaming over a large listening area. It can also handle power, if that's what you want. As for the two 9" woofers in each speaker, they are equal in output level and need no assistance from a subwoofer to yield substantial bass. I'd say that Steve Eberbach's priorities haven't changed much over the years.

My measurements generally confirmed the initial good impression made by the speaker and revealed no important weaknesses. The nominal impedance of 4 ohms specified by the manufacturer is substantially correct in the lower midrange; then the magnitude rises to 10 ohms at 2.3 kHz and comes back again (this with the spectral balance controls at the factory settings). The phase of the impedance is within +307-15°, not a difficult load for a half-decent amplifier. The bass response as measured with the nearfield (Don Keele) technique is that of a 32-Hz box rolling off at 18 dB per octave below the 32-Hz tuning frequency. (That 32 is a reasonably accurate number; I can't swear for the 18 because of the difficulty of summing the rear vent with the forward-firing woofers.) I would call the speaker capable of solid bass response down to 25 Hz or so, quite remarkable for a relatively slender column. Pulsing the woofers revealed a very slightly underdamped condition, probably intentional (extremely tight, lean bass is not the DCM sound). Measuring the inboard (louder) panel head-on with the MLS technique resulted in a 1-meter anechoic response of ±2.5 dB from 300 Hz to 20 kHz (again with the controls at the factory settings). In the tweeter range the response was actually ±1.5 dB. These are excellent curves, needless to say. The claim of superior phase linearity was also substantiated in the MLS test. I was unable to obtain valid results when I tried to measure the ostensibly identical outboard panel because of interference from the much louder inboard panel. The 1-watt/1-meter efficiency is on the order of 89 dB, which is quite high.

I expected to see almost perfect reproduction of gated square pulses by the Time Window Seven, that having been one of the distinguishing characteristics of its 1977 predecessor, but I soon realized that the complex 3-way network wouldn't quite allow it. All drivers are wired in phase and the pulses were reasonably square, distinct, not at all smeared or squooshed, but the leading and trailing edges had big spikes and ripples. The spaces between the gated pulses were free from bulges and other garbage. With any other speaker this would still have been a surprisingly good result. I no longer believe, however, that pulse coherence is audible. On the other hand, storage effects as revealed by tone bursts are audible, but the speaker passed very clean bursts across the spectrum; I saw only one small problem at around 1.4 or 1.5 kHz, nothing serious enough to worry about.

Getting back to the sound, I am sure it is exactly what many people are looking for—smooth, uncolored, refined in texture, unvarying over a wide listening angle, very dynamic on either classical or rock music, panoramic in breadth and depth. The bass may be a tad full for some tastes but even so it's very satisfying. If transparency and focus are more important to you than any of the above, if you like every pore and freckle of the music illuminated, then you may end up preferring a really good monitor-type speaker over the Time Window Seven. I have heard more immediacy, greater you-are-thereness on soprano voice, for example, with a few other speakers, but that's a tradeoff, not a shortcoming. You can't expect all the virtues of a strictly forward-firing speaker in a dual-wave-launch design that has all these other advantages.

I would position the DCM TimeWindow Seven somewhere between the Waveform Mach 7 (monster monitor) and the Carver "Amazing" Platinum Mark IV (super dipole). It doesn't beat the latter two at their own game but it gives you a bit of both worlds in a much more manageable package than either. Nice work, Steve.

Monitor Audio Studio 6

(Reviewed by David Rich)

Monitor Audio USA, P.O. Box 1355, Buffalo, NY 14205. The Studio 6 compact 2-way loudspeaker system, $2499.00 the pair (without stands). Tested samples onloan from distributor.

This is the replacement for the Monitor Audio Studio 10, which was reviewed in the last issue. The new Studio 6 has the same internal volume and the same size drivers as the Studio 10. The vented enclosure of the new speaker is a couple of inches shorter and a couple of inches wider. The new cabinet appears, using the old standby knuckle test, to be as resonant as the old cabinet. The nonstandard binding posts have also been retained. *[Yuck.—Ed.]* Our test unit was finished in an optional black piano finish that would be worthy of a Steinway. The drivers in the Studio 6 have been revised. The phase plug of the woofer of the Studio 10 has been removed. In addition, the protective metal mesh surrounding the tweeter has been removed, exposing the fragile tweeter to potential damage. This tweako move was inspired by John Atkinson, the editor of *Stereophile,* who suggested that the Studio 10 sounded better with the protective mesh removed. I suggest that if you own this speaker and the tweeter is damaged, you should contact Mr. Atkinson for payment of the replacement cost.

The principal problem with the original Studio 10 was a large resonance in the woofer, spanning the range from 3 to 7 kHz. The resonance reached a 5 dB peak at 5.5 kHz. In the Studio 6 the peak now spans a range from 4.5 to 6 kHz and has a maximum value of 2.5 dB. Aside from this difference other measurements are similar to those of the Studio 10. Excluding the peak, the speaker's response easily fits in a ±2 dB window from 300 Hz to 20 kHz. The principal attributes of the response are a broad 3 dB bulge in the range from 500 Hz to 1.5 kHz and a dip in the crossover region between 2 kHz and 4kHz. The magnitude of this dip is strongly dependent on the height of the microphone placement relative to the speaker. This is a consequence of the 2nd-order crossover network used in the speaker. The best results were obtained with the microphone aimed at the tweeter. Horizontal dispersion of the speaker is very good, with little variation at 30° off axis below 13 kHz. The speakers were matched within 0.75 dB. Tone burst response of both drivers was very good overall. A strong ultrasonic resonance was observed in the tweeter beyond the audio range. The 6 kHz resonance was also observable in the woofer. Other smaller resonances were observed in both drivers. Sometimes these resonances corresponded to small amplitude peaks in the speaker's frequency-domain measurements.

The bass response of the Studio 6 rolls off with a -3 dB point at 58 Hz. The response appears to be maximally flat. Distortion at low frequencies becomes high as the cone moves out of the gap. For a high-quality system a subwoofer becomes an almost mandatory addition to the Studio 6. Given this, I believe a more optimal bass alignment would have been a closed box instead of the vented box used. This would allow better matching to a subwoofer and reduce nonlinearity as the speaker attempts to produce frequencies outside its passband. The speaker is easy to drive, since the magnitude of the input impedance never drops below 5.5 ohms and the phase angle never exceeds ±30°.

Sonically, the best features of the Studio 6 have been retained, but the forward character of the Studio 10 is gone. The 6 kHz peak is excited less often because of its smaller bandwidth and is less noticeable when excited because of its reduced amplitude. The Studio 6 sounds open and transparent, provided the height of the speaker is adjusted optimally to eliminate the crossover notch. The speaker disappears in a way that only a small minimonitor emulating a point source can. Instrumental timbre is well-preserved. As was the case with the Studio 10, and in contrast to most audiophile loudspeakers, nonaudiophile recordings are reproduced wonderfully.

The speaker's state-of-the-art sonics are somewhat surprising given its measured performance. Speakers with less advanced driver technology, at less than half the price of this speaker, have similar measured performance—save for the pair matching, which is extraordinary. The cause of the improvement in sonic quality in this speaker, which must be due to its advanced driver technology, is not yet being evidenced in our current measurement regime. Please do not write to tell me a similar situation may exist with electronics. All loudspeakers exhibit many significant measurable deviations from ideal performance. All these deviations are above the limits of audibility. Some deviations, such as frequency response, are clearly very significant. Others, such as linear phase response, appear to be less significant. Still others, such as harmonic distortion, lie between these two extremes. Complicating the process is the fact that different listeners to a loudspeaker will have different sensitivities to each parameter. The question in loudspeaker evaluation is—given a flat frequency response—which of these other deviations, either singly or in combination, have the most impact on audible performance. The designer's job is then to minimize these deviations at the expense of other parameters. Large R & D expenses are incurred at major speaker companies addressing these questions and solutions. As more of this work enters the public domain, our electrical characterization of loudspeakers will more accurately reflect the speakers' sonic performance. Until that happens, it should not be a surprise that a company such as Monitor Audio, which manufactures its own drivers and thus clearly has more control over the optimization process, has produced a speaker with excellent sonic characteristics.

No single speaker can be optimal for all listening rooms. In the Editor's large room the similarly priced Carver "Amazing" Platinum Mark IV loudspeakers moves a lot more air and thus can, in many ways, produce a more realistic presentation. In my small room the Carvers would be a disaster. The Monitor Audio Studio 6 has been designed for small rooms and performed excellently in mine. Overall the speakers performed well enough for me that I decided to purchase them.

Snell Type B Minor

(Reviewed by Peter Aczel)

Snell Acoustics, Inc., 143 Essex Street, Haverhill, MA 01832. Type B Minor floor-standing 3-way loudspeaker system, $3390.00. Tested samples on loan from manufacturer.

The Snell Type B was reviewed in Issue No. 17 and was rated good but not great. Its slightly redesigned junior version for 20% less is a much better speaker in my opinion, probably the best speaker from Snell so far. I'll even assume the risk of calling it great.

What's the difference between the Type B and the B Minor? The latter is a simple 4' high rectangular block, not a pentagonal column. It has one 12" woofer instead of two 10" woofers. It has a different, and apparently improved, front tweeter with a 1" titanium dome; the two 5" midrange drivers, the rearward-firing 1" dome tweeter, and the crossover slopes appear to be the same. The 12" woofer is mounted on the inboard side, the left and right units being mirror images of each other, and that allows the front side of the box to be only 10½" wide, for a minimonitor-like wave launch. The vertical mid/tweet/mid array (sometimes referred to as a D'Appolito arrangement although D'Appolito was not the first to use it) is surrounded by felt, which appears to be every bit as effective as rounded corners for minimizing diffraction. All drivers are wired in phase.

The measurements proved to be outstandingly good. This is one flat sucker. (Help! I'm beginning to sound like Corey Greenberg, even though I hardly ever read him.) The 1-meter anechoic (MLS) response on the tweeter axis, with the tweeter control upright, is ±2 dB from 300 Hz to 20 kHz (except for an additional -1 dB deviation at around 3.3 kHz). I can't remember ever seeing anything much better than that. What's more, at 30° off axis horizontally, toward the listening area, the response is even a little flatter up to 14 kHz or so, with a steep rolloff beyond that. Phase linearity on axis is also outstanding. The response 30° off axis vertically, toward the ceiling, is flat within ±2.5 dB up to 18 kHz except for a huge suckout at 2 kHz, probably a crossover effect and irrelevant to seated listeners in any event.

The woofer in its sealed enclosure rolls of at 12 dB per octave; its -3 dB point appears to be 32 Hz; the pulse profile indicates a Q of 0.7, increasing ("woofing up") a bit as the level is increased and the voice coil begins to leave the gap. The crossover frequency is approximately 250 Hz. Super bass has never been a Snell specialty (at least not in their monolithic full-range speakers); I would rate the B Minor as good but not great on that count. On the other hand, tone bursts are extremely clean at all frequencies. The impedance of the speaker is 4.8 ohms at 200 Hz and rises slowly to 10 ohms at 1 kHz and 13 ohms in the top octave of the audio range. That makes the nominal impedance 8 ohms. The phase angle is within +25°/-10° across the spectrum above 100 Hz, a very easy load for the amplifier. Dual inputs are provided for tweako biwiring—all right, all right, it can't hurt.

The sound of the B Minor is very much to my liking but may be too "analytical" for those who like a softer focus (which they call "musical" and I don't see as such). The new titanium tweeter reproduces the minutest details of the upper three octaves with hairline delineation but without the slightest harshness or overemphasis. I'd love to compare it with the much more costly Accuton and Win tweeters, which would probably beat it but were not available to me at the same time. The midrange of the B Minor is also crystal clear, showing no trace of the "subtle coloration or lack of ease or stuffed-up quality" that baffled me so much in the Type B, although the 5" midrange drivers appear to be the same, as I said—but without the tone-burst irregularities of the B. Maybe that was some kind of interference effect due to the cabinet, not a storage phenomenon—I'll never know. The entire sonic presentation of the B Minor is what a monitor-type speaker ought to give you but seldom does—in-your-face presence and detail combined with excellent octave-to-octave balance and nothing sticking out, nothing annoying. It's not as big and room-filling a sound as that of the Waveform, the Carver "Amazing," or the DCM Time Window Seven, for example, but it's a very clean, wide-open window on the program material.

The bass is obviously deep and clean but doesn't quite have the impact I look for in this price range. In a smaller room than mine that may turn out to be less of an issue. I believe this minor limitation is due to the off-the-shelf (though clearly high-quality) 12" driver; great bass in an enclosure of this size can be achieved only with a custom-designed driver. I was able to find an excellent solution simply because I have all kinds of amplifiers and electronic crossovers in my laboratory on extended loan: I added a pair of Hsu Research HRSW10 sub woofers to the Snell B Minors, fully biamped and crossed over at 40 Hz with 18 dB per octave slopes. The feed to the subwoofer amplifier is set 4 dB higher than to the main amplifier. Wow! That's a sound I could live with for a long time; it borders on the awesome; but the allocation of dollars for each component of the system doesn't make any sense to the audiophile who would have to pay for it all. Even if used straight out of the carton, however, the Snell B Minor is a—why not say it?—great speaker. •

A First Look at Perceptual Coding and the Digital Compact Cassette (DCC)

By Peter Aczel
Editor and Publisher

Here is a situation where you begin to wonder whether the whole isn't smaller than the sum of its high-tech parts.

I would have preferred to report here on the Digital Compact Cassette versus the MiniDisc (MD) head-to-head, but our Sony MD recorder sample arrived much too late for that. The next issue will definitely include a review of the MD technology.

Marantz DD-92

Marantz USA, a Division of Bang & Olufsen of America, Inc., 1150 Feehanville Drive, Mount Prospect, IL 60056. Model DD-92 Digital Compact Cassette Deck, $1200.00. Tested sample on loan from manufacturer.

I think it will be best if I state my basic position on the subject of DCC up front, before any discussion of our tests of the Marantz DD-92. This is essentially a political product, not a technological one. (The same is true of all other DCC decks, needless to say.) DCC would never have happened if R-DAT had not run into serious political trouble in the consumer market. R-DAT is small, cute, and cuddly; it is versatile and practical; its audio fidelity is indisputably state-of-the-art; its linear PCM coding is uncompromised; it is the perfect digital recording/playback medium for our time. DCC is bigger, cruder, clumsier, much less lovable; its operation depends on low bit-rate coding, which is still being hotly debated as to audio quality; its backward compatibility with the Philips audio cassette—for playback only!—is of small consequence because everybody has at least one cheap cassette deck for that purpose (not to mention making copies for the car stereo, which a DCC deck cannot do). R-DAT blows away DCC in every way, but the marketeers got tired of fighting the paranoid political opposition to the perfect recording/copying medium and abandoned the consumer market (though not the professional market). DCC is what tries to fill the resulting vacuum; it is claimed to be audibly perfect while circumventing bit-for-bit copying of linear PCM recordings.

It has been argued that R-DAT would have remained prohibitively expensive for the mass market even if there had been no opposition to it. That argument is contradicted by the universal abundance of cheap VCRs, based on the same sophisticated rotary-head technology as R-DAT. The irony is that DCC may never come down in price to the mass market level because, according to recent reports, the manufacturing costs of the supposedly simpler DCC magnetic head have been disastrously underestimated. It would be poetic justice if DCC failed and consumers started to demand cheaper R-DATs.

That said I must now state that the Marantz DD-92 is in many ways an admirable piece of engineering, proving what can be done when a resourceful industrial giant, namely Philips, decides to do an end run around a head-on unyielding technological/political problem. I am quite impressed—with a few reservations.

The basic idea behind low bit-rate coding—more illuminatively called perceptual coding—is that you can't possibly hear a mosquito circling around a jackhammer in action, so why not record only the jackhammer and use fewer bits? The assumption is that only so many bits will fit into so many seconds on the medium (tape, laser disc, etc.), which may not be true as the media technology advances but is a practical fact for here and now. The trick is to capture absolutely everything that the keenest ears can hear, no matter what the musical or other program material happens to be, and to throw away only what they surely cannot hear. There are a number of rival coding schemes purporting to do exactly that; the Philips algorithm is called PASC (Precision Adaptive Subband Coding) and uses 4-to-l data reduction. By far the best and most complete explanation of the system appeared in the September 1991 issue of *Audio*, by none other than our own Contributing Editor at Large, David Ranada. My congenital reluctance to do once again what somebody else has already done with great skill has been frequently reiterated here, so I shall simply refer the reader to that excellent article and blithely proceed with my own evaluation.

The Marantz DD-92 is the flagship DCC deck of the Philips line and looks it. Cosmetically the unit is in the top-of-the-line Marantz idiom, meaning the massive gold look with thick, crackle-finished end caps and a smooth, slightly curved fascia, plus lots of bells and whistles. Only the remote control unit looks a bit chintzy. The cassettes are the same size as the ordinary ones but a little more high-tech-looking and equipped with a sliding metal shutter like a 3½" computer diskette. The microprocessor-controlled displays in the front-panel window are amazing—they tell you in real time everything that's going on, everything you need to know, and a few things you don't need to. I'd say that those who judge audio equipment by the controls will be happy.

The most important thing I wanted to find out about the unit was whether or not PASC data reduction resulted in any *audible* change in the signal. David L. Clark, whom I have always found to be reliable, had already reported in the April 1992 *Audio* that PASC was audibly transparent, so I was at least emotionally prepared to find likewise. I copied the entire 33-minute length of the Stravinsky *Sacre* (Levi/Atlanta on Telarc, Grammy nominee for engineering) from CD to DCC via S/PDIF; then I also copied the Bach Toccata in F-sharp Minor, BWV 910, for harpsichord (Colin Tilney on Dorian) onto the same tape, as I did an undistinguished but superbly recorded jazz cut from a *dmp* CD. The Stravinsky has very dynamic, heavily orchestrated passages alternating with soft, delicate ones using only a few instruments; the harpsichord is supposed to be especially difficult for low bit-rate coders to reproduce; the jazz selection contained percussive sounds with clean attack/decay against a quiet background. I set up the ABX double-blind comparator with the CD player and the DCC deck as the signal sources; I matched the levels very carefully and synchronized the CD with the tape for each comparison. Three different experienced listeners—I was one of them—spent about two hours each trying to hear differences. Each of us believed to have zeroed in on very subtle differences as long as A and B were known; in the X tests, however, we got completely random results. Based on this admittedly limited experiment, DCC appears to be transparent.

I understand that David Ranada, who years ago found precisely the right music to identify the sonic Achilles' heel of the unlamented Copycode, is trying to do the same with DCC and other perceptual coding schemes. It stands to reason that 4-to-l data reduction should throw away some audible music now and then, even if very rarely, since the algorithm is most unlikely to anticipate every possible musical combination in the world. So far, however, no one has come up with the music that will confound PASC, at least not in print with documentation. It should be added that this is not a trivial, hairsplitting, audio-freak issue; perceptual coding in some final standardized form will definitely be needed for the data-intensive multichannel audio formats of the future (see the Mark Davis interview in our last issue) and for digital broadcasting; there is no way around it. We must find the all-around best code or face another stupid format war, which could become the most counterproductive in the history of the industry. AT&T claims they have something significantly better than Philips, Sony, and the rest; the silicon implementation, however, may turn out to be costly. Dolby also has a coder. Eventually there will have to be a showdown and a shakeout.

After our listening tests, I decided that it made little sense to test the operation of the PASC, although Audio Precision has a complete test protocol to do so on the "System One Dual Domain." The coder drastically alters the signal, and I was not particularly interested in electronically tracking the procedure as long as it turned out to be sonically transparent. On the other hand, I was interested in testing the A/D and D/A converters used in the Marantz, as they are supposed to preserve the integrity of the signal in their conversions.

The A/D converter turned out to be quite good but not state-of-the-art. Its full-scale THD + N averaged approximately -91 dB across the audio spectrum, showing little fluctuation with frequency. That's just a tad short of 15 bits; Marantz calls it a Bitstream Sigma-Delta converter with 18-bit resolution. Ahem... The linearity of the ADC was excellent, departing from the 0 dB line only below the -80 dB level; the error at -90 dB was +0.5 dB.

The D/A converter performance was not nearly as good—and this is the top-of-the-line Philips Bitstream DAC 7 in the differential mode. Full-scale THD + N versus frequency stays between -88 dB and -93 dB from 20 Hz to 2 kHz, then goes into orbit: -79.5 dB at 8 kHz, -64 dB at 18 kHz! No it isn't jitter because with the S/PDIF input reduced to -20 dB it goes away; normalized to full scale it then looks like a -95 dB DAC up to 10 kHz, and the normalized 18 kHz distortion is also a much more respectable -84 dB. This is gain-related analog distortion, almost surely, and it isn't necessary. Remember, there are DACs that can give you that -95 dB performance at full scale and at all frequencies. Len Feldman, reviewing the Marantz DD-92 in the March 1993 issue of *Audio,* shows pretty much the same curve I obtained but leaves it without the slightest criticism. Bob Harley in the July 1993 *Stereophile* doesn't even mention full-scale THD + N versus frequency in his review of the DD-92; at this point I think he avoids that measurement because I keep insisting on it. (Commercial: Read *The Audio Critic* and find out what the other reviewers will never tell you.) After that I must add that the gain linearity of the DAC was superb, requiring no critique.

To sum up, the Marantz DD-92 delivers the sonic performance claimed for it and is overall a pretty sexy toy from the point of view of the average knob-twiddling audiophile, but it has some measurable shortcomings that aren't easily forgiven at the $1200 price. For $1200 I'm sure that Philips with their present-day technology could produce a DAT deck that would need no apology and no convoluted political reasoning behind it. On the other hand, if PASC turns out to be the perceptual coding scheme of the future—frankly, I doubt it—then the total effort will have been more than worthwhile, and the DD-92 will be remembered as an industry milestone. •

Top-of-the-Line Pioneer Elite Video Equipment

By Peter Aczel
Editor and Publisher

How good can rear-projection TV and laser videodisc playback be these days? Awfully good, and in some cases unnecessarily costly.

I remember muttering something to the effect that there would be one high-end video review per issue—to test the waters of the home theater scene—and here are two, or at least one and a half. The idea is not to sidetrack, and certainly not to bore, a primarily audio-oriented readership but to keep an eye on the home entertainment industry's emerging mainstream, which has obvious impact on the direction audio is currently taking. Audio and video are no longer two worlds, that's quite clear.

55" Rear-Projection TV

Pioneer Elite PRO-106

Pioneer Electronics (USA) Inc., 2265 East 220th Street, P.O. Box 1720, Long Beach, CA 90801-1720. Elite PRO-106 Reference Projection Monitor Receiver, $4500.00. Tested sample on loan from manufacturer.

Once again, I'm reviewing a TV set near the end of its life span; by the time this is in print the new Pioneer Elite PRO-107, which replaces the PRO-106, will be only weeks away from its debut. This can't be helped; TV models change too often and are not always available to reviewers at the very beginning of their commercial life cycle. It doesn't matter a great deal in this particular case because the similarities between the PRO-106 and PRO-107 appear to be much greater than the differences; the biggest difference will be the $1000 higher price of the latter! That alone may cause prospective buyers to shop for the older model, possibly at a discount.

Overall, this is the most impressive TV I have lived with for an extended period of time. (I'm not counting my experiences with front-projection TV, which is not suitable for typical domestic environments; see also Issue No. 19, page 37.) The screen of the PRO-106 is 3" bigger diagonally, and therefore 12% bigger in area, than that of the 52" Magnavox (Philips) rear-projection TV I reviewed in the last issue—and it makes a difference. The impact is that much greater. For movies (especially when "letterboxed"), sports, opera, travelogues—I could go on—*big* is more important to me than *sharp,* in the sense that I am willing to give up a little resolution to gain greater size for a more lifelike effect. Others could legitimately disagree with me. Not that the PRO-106 is lacking in resolution, far from it. It's just that the picture size impresses me even more than the excellent picture quality. Add to that the beautiful black lacquer finish of the unit, which somehow streamlines and minimizes its gigantic dimensions, plus the smoothly gliding casters that make it easily movable on any surface except a thick carpet, and you have a very good first impression.

The advanced technical features of the PRO-106 are numerous but not particularly meaningful to most readers of an audio magazine without a video tutorial. (One of these days we'll bring you one.) Pioneer seems especially proud of the automatic digital convergence system, the high-contrast lenticular "black screen" with 0.9 mm pitch, the f/0.98 short-focal-length high-luminance lens system, and the 3-line digital comb filter. The exact benefits of these and other features are discussed in the detailed "Technical Notes" that come with the set. Of greater immediate concern to the user is the separately packaged transparent acrylic panel that can be installed over the screen at the user's option; the set is designed to be used with or without it. I decided to install it. It not only protects the lenticules against dust and scratches but also makes the scan lines unnoticeable to the viewer; you could call it a kind of optical dither. Quite frankly, this simple feature may be the only reason why I found the picture of the Pioneer to be subjectively somewhat superior to that of the aforementioned Magnavox, since the test patterns I used gave no clear indication of such superiority. On the other hand, the four built-in 25-watt amplifier channels and two JBL speaker systems of the Magnavox totally outclass the two 10-watt channels and dinky little tucked-under speakers of the Pioneer. I think Pioneer would much rather have you connect the PRO-106 to an external Pioneer Elite audio system, especially one with Dolby Pro Logic facilities, which the set does not include (but the Magnavox/Philips competition does).

What about the microprocessor-controlled razzle-dazzle expected of an expensive TV these days? The PRO-106 has all the menu-driven audio/video adjustments and conveniences you're likely to look for, and then some. One of the "then somes" is an extremely versatile picture-in-picture (PIP) function with a multiscreen feature. You can select a 4-subpicture or a 9-subpicture mode; you can even scan 9 different channels as still pictures to get a quick idea of what's going on. The younger generation in my household considered this feature to be particularly "cool." (Of course, to see two ongoing programs in real time as a main picture and a subpicture, you need two sources, such as a VCR plus the TV.) On the debit side, when I tried to label each channel with the channel number plus the station call letters, as I had done on the Magnavox, I found that it couldn't be done—it was one or the other label on the screen but not both. So the microprocessor capability of the set is considerable but not the ultimate. I did find the remote control unit to be ergonomically superior to that of the Magnavox.

My tests with the Reference Recordings laser videodisc *A Video Standard* yielded mixed results. The picture adjustments as programmed on the PRO-106 have a relatively narrow range on the plus side of the basically well-chosen default settings; when you want *more* of something, not much happens. Black level retention—which is the ability to hold black at black, independently of the picture content—was neither surprisingly good nor worse than what is considered basically satisfactory in anything but a professional monitor. Contrast could not be turned up to the point where the peak linear capability of the set was exceeded but was judged quite sufficient in the available range. Color performance via the S-video input resisted adjustment to the test disc's ideal settings, but I had no quarrel with the PRO-106's default color and tint adjustments, which inclined perhaps toward the cool side. Geometry was very precise; checkerboard patterns and circles were reproduced without distortion. The original factory setting of convergence required no trimming during the long months the set was in use. The advertised 830-line horizontal resolution of the set (yes, eight three oh!) could not be verified, of course—the industry is totally unregulated when it comes to this spec—but the 425-line resolution of my test setup was to all appearances equaled or surpassed, not to mention the 336-line resolution of NTSC broadcasts (best case). The set has a VNR (Video Noise Reduction) system intended to reduce the video-noise content of typical TV programs and prerecorded videotapes; I found that the resolution of the set was distinctly better with the VNR turned off.

One pleasant surprise was the viewing angle—everyone who had also viewed the 52" Magnavox commented on how much more accommodating the Pioneer was in this respect. You can sit quite far to the side.

My overall assessment of the PRO-106 is that its picture quality and user features are right up there with the best 52" rear-projection TVs and that its 55" size therefore puts it at the head of the pack—except possibly for the forthcoming PRO-107, which has some untried new digital picture-enhancement features, for a lot more money. But don't look for audio quality in the PRO-106; it needs an external audio system for home-theater use.

Laser Videodisc Player

Pioneer Elite LD-S2

Pioneer Electronics (USA) Inc., 2265 East 220th Street, P.O. Box 1720, Long Beach, CA 90801-1720. Elite LD-S2 Reference LaserDisc Player, $3500.00. Tested sample on loan from manufacturer.

Why is this unit is being reviewed here as the companion piece to the Pioneer Elite PRO-106 rear-projection TV? Well, it's almost as expensive and it's cosmetically a matching accessory—I wish there were better reasons. This is basically a "statement" product: the costliest and most uncompromising design Pioneer has been able to come up with for videodisc playback exclusively. The LD-S2 won't even play CDs, only the various types of CAV and CLV videodiscs in the 12" and 8" sizes. I can honestly say that in actual use the LD-S2 gives no more pleasure visually and sonically, and is no more fun to operate, than the Philips CDV488 universal videodisc/CD player, which I reviewed in Issue No. 14 and which then cost $1300. The LD-S2 is of course an incomparably more beautiful piece of machinery—it had better be!—but is that worth $2200 extra? To some it is, undoubtedly. Why? "Because it's there."

For openers, the LD-S2 weighs 6½ pounds. Not many high-wattage power amplifiers weigh that much. The chassis is a massive honeycomb casting with graphite damping. The laser pickup with its linear drive system and the spindle motor are gorgeous precision mechanisms mounted on a diecast aluminum subchassis. Highly elaborate floating/damping/silencing devices are employed throughout. The demodulated image signal is processed digitally via a sophisticated DSP circuit; this is claimed to realize optimum phase characteristics in the color signal output. Both coax and S-video outputs are provided, of course. The remote control facilities are versatile and easy to use, but any number of recent laser videodisc players at a fraction of the price have comparable picture-manipulating capabilities. The LD-S2 is several years old, and microprocessor chips are cheap these days.

I tested the unit with the applicable test signals on the Reference Recordings laser videodisc *A Video Standard* to verify as many of the specs as I could. The 425-line horizontal resolution claimed appears to be correct. The digital audio section's claimed dynamic range of 100 dB is exaggerated; I measured 90.5 dB in one channel and 88 dB in the other. The claimed 0.0015% THD of the digital audio section is also overoptimistic; I measured c. 0.003% (i.e., -90 dB) across the audio spectrum at full scale. That's fairly respectable even for a high-end CD player but not state-of-the-art. (Maybe their spec does not include noise; that could account for the difference.)

Bottom line: an impressive piece of precision machinery that I would be happy to receive as a gift but would buy only if it included CD and cost $2000 less. •

Outboard D/A Converter Followup: Upgrades by Enlightened Audio Designs

By Peter Aczel
Editor and Publisher

EAD is looking good as they keep doing their thing, which is the steady refinement of the multibit approach to D/A conversion.

For background information, you are referred to the EAD reviews in Issues No. 16, 17, and 19. This is one of the truly "enlightened" companies in the digital field, and we have been following their work with particular interest.

EAD DSP-7000 Series II and DSP-1000 Series II

Enlightened Audio Designs Corp., 300 West Lowe, Fairfield, IA 52556. DSP-7000 Series II outboard D/A converter, $1995.00; DSP-1000 Series II outboard D/A converter, $999.00. (Prices include optional AT&T glass optical input but not optional balanced outputs.) Tested samples on loan from manufacturer.

In my last EAD review, I already mentioned that they were about to switch from the 20-bit Analog Devices DAC in the DSP-7000 to the 20-bit Burr-Brown PCM63P-K, which is very near the top of the heap in multibit DAC chips. Since then EAD has made available this so-called Series II upgrade both as a kit for installation in the older units and as a standard part of the current line. That line now includes the DSP-1000 Series II, a remarkable value as we shall see.

The Burr-Brown DAC has a higher current output than the AD, and that has created a bit of confusion. The early Series II versions of both models, in which the Burr-Brown chip was a simple plug-in substitution, had an unusually high line output of 4.7 V with a full-scale digital input. EAD thought that was a good thing—great for users of tweako passive control units, for example—but then contrary opinions started to come in, and they came up with a better solution. The latest Series II production units come out of the box with 2 V standard line output, quickly modifiable by the user to 4.7 V if so desired. My test samples still had the 4.7 V output only, which appeared to produce a very small amount of gain-related analog distortion, so in my THD + N versus frequency tests I reduced the full-scale (0 dB) digital input to approximately -7.4 dB to obtain a 2 V line output, then I corrected the results by 7.4 dB to normalize them to the later standard configuration. I believe this method yields realistic figures, subject to confirmation when I get my hands on later production units.

With this method I found that, in terms of THD + N, the upgraded DSP-7000 shows only a minuscule improvement: in Issue No. 19 I reported a tiny residual amount of gain-related analog distortion rising with frequency; now it no longer rises with frequency (except at the full 4.7 V output). That's all. The main improvement is in gain linearity—it is now absolutely perfect, with +0.2 dB error at the -100 dB level, beating delta-sigma converters at their own game. I've never seen anything as good, and the low-level linearity is confirmed by the squeaky-clean spectrum of a dithered 1 kHz tone at -90.31 dB.

After that, I was really curious to see what EAD could with the same great Burr-Brown DAC in a stripped-down processor costing half as much. The DSP-1000 Series II is a modest-looking but far from unattractive pancake-style box incorporating many but not all features of the DSP-7000 Series II (fewer front-panel indicator lights, an obviously less hefty power supply, etc.). To my great surprise, there was no comedown in measurable performance whatsoever! The measurements of the two models were in effect interchangeable—I am not exaggerating!—except for the 120 Hz bump in the noise floor of the 1000, a consequence of the power-supply compromises. At its -110 dB level, however, that bump is hardly a performance consideration. If there was any other difference in the measurements, it was very slightly in favor of the 1000 (for example, in channel separation, if the difference between 107 dB and 100 dB at 1 kHz means anything). I found myself hard put to find a good reason for spending $996 extra on the purchase of a 7000. I asked David Rich what he thought of all this and he sent me the following note:

"An improved power supply alone would not entice me to come up with the extra cash for the DSP-7000 Series II. For the added funds I would want to see an S/PDIF decoder with better jitter performance than can be obtained with the Crystal chip alone. For example, Crystal Semiconductor has produced an application note on how the jitter performance of the CS8412 can be significantly improved by adding to the S/PDIF decoder a PLL (phase-locked loop) that uses a VCXO (crystal-based voltage-controlled oscillator). I am also surprised, in view of the fact that EAD now produces a CD trans-

From Tweak to Weasel
(in One Easy Step)

By Tom Nousaine

Editor's Note: *Tom Nousaine has decided that the* geek *he so carefully defined in his column in the last issue should be spelled* geak, *to emphasize the juxtaposition to, and contrast with,* tweak. *I am not against creative misspelling as long as it is deliberate and reasoned, so I am not correcting it. There may even be some advantage to distinguishing an antitweak audio geak from a nerdy college geek or a circus geek who bites off the heads of live chickens and snakes.*

* * *

Well, I initially wanted to use this issue to outline the equipment needs of a hard-core High Definition Audiophile, aka Geak, but a happenstance at the Stereophile High-End Show in San Francisco made me decide to, instead, cover the backlash you will get from the High End when you employ rational thinking.

So, I was browsing through exhibit rooms at the show and happened upon Clark Johnsen, the self-proclaimed absolute-polarity hero of the '90s. Clark, upon recognizing me, started right in by asking if I "believed" in absolute polarity. I responded that I had no evidence to support the notion that absolute polarity was of any significance to home audio enjoyment.

Clark kept right on. He immediately claimed that Stan Lipshitz had shown that absolute polarity was audible on reproduced music. I responded that I had been there when Stanley initially collected data and that the results were significant only when trials using test tones were included in the analysis.

Clark moved right to Dick Greiner's experiments on polarity. My response was that I had personally discussed that work with Greiner, and it turns out that he was able to define special circumstances with special test tones (resembling certain trombone notes) where subjects could reliably hear polarity, but ultimately it was of no, or extremely minor, significance in the playback of recorded material in the home.

Therefore I concluded Clark's position that absolute polarity is of major importance and immediately recognizable on many recordings was just not supported by any scientific evidence. Especially not his goofy AES "paper" with the spurious "triple-blind test."

Soon we had an "experience." Clark trotted out a recording to demonstrate his point. We played a short excerpt. Replayed it. The audience asked for an encore. I asked for another. Afterwards Clark asked, "Anyone hear a difference?" The guy next to me looked over and shrugged. I shrugged back. He then looked over his shoulder and saw some raised hands. After this he raised his hand about halfway to Clark's appreciative smile.

Clark shouts: "There you have it, 6 out of 6 heard a difference!" I responded, "I didn't raise my hand, Clark." He retorts, "Oh well, 6 out of 7 then." Interestingly, there were more than 8 people in the room (more than 6 plus me and Clark). I guess Clark just didn't bother to count hands that weren't raised. So much for Johnsen's experimental results.

I said that I heard differences on all four trials. And I did; as I often do the first several times I hear a new tune. As all people will do at least 3/4 of the time, even without coaching, when given identical alternatives.

Clark inferred that people in the room heard absolute polarity. Maybe they did, but you can't tell from this session. First, Clark only asked if anyone heard "a difference." He didn't mention the word "polarity" at any time during or after the actual listening session. Maybe some heard the drummer's technique more clearly the second time. Maybe some of them were impressed by the power of the vocal when they heard it again. There were no controls of any kind; and no reason to conclude that absolute polarity differences were being heard.

What's the point? If you are going to be so bold as not to hear what your coach tells you to hear, you will have to pay. My penance was to have Clark Johnsen label me "a weasel" in his incomplete and misleading *Ster-*

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port, that the DSP-7000 does not allow for a two-wire data communication system. Such a system—used by Krell, Sumo, Denon, Linn, and Deltec among others—eliminates the S/PDIF jitter problem. In addition, I would want a more advanced digital filter than the NPC SM5813. Filters such as the NPC SM5842AP and the latest filter from Sony, the CXD2567, have dithered multipliers and wider data paths than the SM5813."

Far be it from me to question David Rich's judgment when it comes to silicon, especially since I came to pretty much the same general conclusion on my own. The big value here is the EAD DSP-1000 Series II, and I am currently recommending it to all those who ask me about outboard D/A converters.

Erratum by David Rich:

In Issue No. 19, page 42, second column, tenth line from the bottom, the sentence reads: "The division process reduces jitter from the VCO." This is incorrect. I should written the following:

A clock must have a maximum jitter significantly smaller than its period. A very high-speed VCO with an output of, say, 10 GHz center frequency, must thus have jitter which is much less than its 100 ps period. Properly done, the division process would introduce no additional jitter.

Reducing this to a single sentence: The division process preserves the low jitter of the VCO.

—David Rich

eophile recount. Editor John Atkinson footnoted me as the kind of person who "rubbishes the statistically significant results" about phenomena "he apparently believes to be inaudible." *[Do Brits use* rubbish *as a verb? Not the ones I associate with.—Ed.]*

Can this be true? Nousaine ignores or discounts contrary evidence? Of course not. That's what Atkinson and Johnsen do. No one has proven that absolute polarity is audible on reproduced playback of music. Greiner suggests that a recording standard concerning polarity might be a good idea. But he comes nowhere close to suggesting you can hear it in your home, or in Clark's store, or in John Atkinson's reference system, except with the special trombone-like test signals used in the experiments.

The lesson: the truth makes many people angry. Especially audiophiles. Many in the high-end audio community have more than just a technical aversion to negative feedback. When the truth is politically incorrect it will be pilloried, and you along with it if you're standing too close, by those whom it offends.

Also be wary of those situations where you may "mistakenly" fail to hear differences. You probably recognize most of them from experience. First, the Prophet arranges a listening session, and you identify the wrong choice as best; this is repeated with "better" program material until you give the right answer. This can be done very subtly and in an extremely convincing way. Do not ignore the obvious. Why didn't he use the best material first, and why didn't you hear it the first time?

Another danger signal is impatience or indignation. I once had a Linn Sondek salesman rudely attack my "insensitivity" and "confusion about accuracy" when I "mistakenly" selected a Linn record playback system as being inferior to CD. I was summarily escorted out of the store with instructions not to return until I had overcome my confusion.

Raise the red flag when the Prophet tries to negotiate differences with you. I recently was invited to a High Ender's home where he spent a great deal of time telling me about the wonders of modifications made to his electronics and listening room. He played a few selections that were familiar to me, after which I told him what I was hearing. When this differed from expectation he tried to negotiate with me. He actually wanted me to agree to hear X even if I couldn't hear Y. He was trying to cut his losses.

This is such a shame. As is often the case, this guy was really after confirmation, not another opinion. Consider carefully what happened here. When you ask someone for an opinion, try not to comment on it. Listen carefully to exactly what the subject says. Interpret it literally; think about it; use the information as data. Don't ignore, reinterpret, or dismiss the feedback. A good way to get useful feedback is to ask for written, private responses which can be analyzed later.

Clark Johnsen also implied that other respected researchers, Lipshitz and Greiner, either supported his work or had conducted experiments that supported his work. This is, of course, simply not the case. This is even worse than the phony testimonial ("Michael Jordan uses XYZ cable") so often used in other advertising today.

Watch out for this trick. It is very hard to defend against unless you have access to the papers themselves. Be forewarned that no one has shown that polarity flipping, chassis clamping, changing amplifiers or preamplifiers or wires or D/A converters, component break-in, line conditioners, or hanging diapers in the corners of your room can have any possible positive effect on stereo reproduction except in the case of poorly designed products. *[Or good products unsuited to their particular application, such as a low-powered amplifier driving a low-efficiency speaker in a large room.—Ed.]*

Another regularly encountered "encounter" occurs when a group gets together and forms a consensus on what they hear. Ever been there? Everybody sitting around. The host pops in a CD and afterwards casually asks, "What do you think?" At first, everybody pipes in together with a bunch of different stuff, none of which you heard. Then, after a replay, people start to hear more and more of the same stuff. Ultimately a concensus is reached, and you all now hear what you agreed to hear.

This is all nice and comfortable but is really only a different form of negotiation, not that dissimilar from the other case. There is nothing really wrong with it except it is not an acceptable means of determining what people were truly able to hear. For that, you need private written responses, careful selection and repetition of program material, scientifically random ordering of presentations, experimental controls, and rational analysis of data.

Scientific listening is hard work. You cannot just "put bias out of your mind" because you probably aren't even fully aware of your own. Even professionals can't. That's why the smart ones put their theories to test with double-blind listening tests. A High Definition Geak knows that anecdotal evidence and casual demonstrations may signal the need for controlled listening tests to determine the audibility of a phenomenon.

But he also knows that factors that truly make a difference can always be identified with controlled listening tests. Those that can't be verified when the blinders come out or the coach leaves the room are best left to those for whom status, high prices, conformity, and brand names take priority over performance. Do not waste resources on areas where differences cannot be verified. •

Interviewing the Best Interviewees in Audio Part III

By David Ranada
Contributing Editor at Large

We have only a single interview in this installment but it's a good one, with one of the few genuinely original (and realistic) thinkers in the audio industry.

Editor's Note: *This interview was taped approximately a year and a half ago, but there's nothing in it that isn't still timely. What has happened meanwhile, and what would surely be discussed in the interview if it were taking place today, is that Ken Kantor has come out with his no-compromise "statement" in loudspeaker design, the NHT Model 3.3, a 4-way system retailing for $4000 the pair. A review sample has been promised to* ***The Audio Critic.***

7. Interview with Ken Kantor, Speaker Designer and Research Director

RANADA: How did you get interested in audio?

KANTOR: As it was with a lot of people my age, music came first. I was both a consumer of music and also a young musician, and I guess I gradually became interested in hi-fi, first as simply a means of getting music. I wasn't an issue of quality of reproduction; it was simply an issue of "gee, if I build that I can have it." My first real hi-fi system was an inherited old AR turntable that was kind of broken and some old vacuum-tube equipment. I guess I got interested in making it work. Gradually I began to get addicted to better quality. I caught that bug: you hear a good system and it's hard to go back to a less good system.

RANADA: How old were you when this all started?

KANTOR: Well, my very first experience with hi-fi was at about seven. My dad was a technical type, a chemist with a technical background. I remember him explaining the basics of electronics to me. Two of the things he explained were that amplifiers were there to make the signal bigger, and he also explained that resistors were used to impede the flow of electricity—they were like valves. So one very long, boring summer day I was hanging out at home and I had nothing better to do. The inspiration struck me that these two things, resistors and amplifiers, were at cross purposes, and that if I went inside his amplifier and clipped out all the resistors it would probably do a much better job of amplifying. It didn't really occur to me in my seven-year-old cognitive state why it was that nobody else had figured out what I had. But, after about an hour of whining, I talked my mother into letting me have access to my father's amplifier: "Dad said it was OK." You know how kids can be. I actually did it. Needless to say, when he plugged it in to use it, it was in desperate shape. He had a little pile of resistors and a smoking heap. That was my first introduction to audio design. There was a lapse after that. I got back into music in junior high school, playing guitar and playing around with audio effects and audio equipment. I became, as a lot of serious hobbyists are, knowledgeable of the work that was going on in the field, about what made a product sound good. I subscribed to the mainstream scientific thought: a loudspeaker should be flat, et cetera. One day I was visiting a friend who lived in a different city and I wandered into a high-end store, probably around 1970. I heard high-end audio equipment, and it was a revelation. It sounded so different, in some ways so much more accurate than what I had been listening to that it inspired a quest in me that eventually became my thesis project at MIT and eventually a large part of the work I did at AR—what in the conventional set of measurements we perform on loudspeakers is being overlooked? What I heard in the high-end stores were a lot of dipoles and alternative technologies. There were certainly flaws in their reproduction, but they did some things so much better than the conventional speakers I had been listening to, from the point of view of imaging and detail. I really became quite intrigued with the issue of psychoacoustics. That is, what perceptual issues were being overlooked by the technical approach to the transfer function of a loudspeaker?

RANADA: A typical response when exposed to high-end equipment is to become more equipment- and less measurement-oriented. Why did you take a different path?

KANTOR: I guess there were two reasons. The major influences on me were always rational, and it just never occurred to me to do anything else. I had had a rationalist scientific upbringing and I had a little bit of exposure to psychology. And it seemed to me quite logical that the issues of perception and psychology and psychoacoustics were the only ways to look at these questions. I'm a curious person, and mystical approaches don't get you anywhere in terms of curiosity. Going back to the seven-year-old story, that was a curiosity-driven event. I want to see why things work the way they do. I want to make them better. A mystical approach or, for that matter, any heuristic approach to equipment may yield good designs and good-sounding products, but it doesn't yield a lot of knowledge about how to generalize that or duplicate it. That, to me, was never compelling. It wasn't a conscious decision; it just seemed that the way I had to approach it was to understand what was different about what I was hearing.

RANADA: You went to study psychoacoustics at MIT. What was your thesis work on?
KANTOR: My undergraduate work was in the department of electrical engineering. The story gets a little more complicated, but—for the sake of brevity—I was not working on an audio-reproduction thesis. I was working on the synthesis of sound. I had sort of lost interest in equipment as opposed to perception. But, somewhat randomly, I was assigned to a lab facility where there was a lot of very interesting work and equipment. And I found it harder and harder to stick to my assigned topic. I found the work that was going on in loudspeaker design. It was, in fact, the lab where Bose was doing work. My thesis was entitled "A Psychoacoustically Optimized Loudspeaker" or something like that. It dealt with a new approach to loudspeaker radiation patterns, which attempted to reconcile what I perceive and understand of the performance benefits of different types of loudspeakers.
RANADA: So your hi-fi-oriented work started in college?

KANTOR: Oh, certainly. I got this idea for a new way to build a speaker. It wasn't a very practical way but it was an interesting way. It was similar to the work I did at AR with the MGC-1 speaker. It was a user-controlled-directivity, nondelayed system combined with a delayed ambience system. But I needed support for my thesis work, supplies and parts. One of the people in the lab suggested that AR may be interested in supporting this kind of work. So I wrote a very shy one-page letter to Bob Berkowitz [who was then at AR] saying, "This is my idea; would you be interested in supporting this?" Lo and behold, I got back a letter with a $200 check and a "come on out and visit" and "we'll give you woofers and tweeters and cabinets and whatever you need." I can't tell you how exciting that was to an undergraduate student. Two hundred dollars was like a semester's worth of meals for me. That was the big time, to go out there [to AR] and use the facilities and get drivers there for free. Now, of course, I realize that $200 is like a nice lunch with a couple of magazine folks, but it seemed very generous at the time. And that was the beginning of my professional association with audio. It sort of happened accidentally, it grew out of my curiosity. That launched me into a domain where I started building a lot of laboratory prototypes and lab equipment. It also got me really interested in psychoacoustics. How could I relate my own work to this body of academic work on perception? I began to do design work for other companies while still an undergraduate. When I got out of school I got a letter from NAD in London saying, "We heard about some of your work, and would you like to come over and work in our group here?" They didn't have to ask twice. I packed up and moved to London. I worked there for a while designing some early computerized test systems and electronics (a phono stage and tone control). That was my first real job, one confronting issues of mass production.
RANADA: Could you go through some of those? There are many things the consumer doesn't think about when he buys and amplifier.
KANTOR: There are many, many designs that can work when you build one in a lab. For example some designs are not tolerant of the kind of variations that you find when you have to buy 25,000 transistors. How do you build a circuit that really gives repeatable performance over the kind of variations you see in the real world? That's a real issue. It's the way it is with amateur loudspeaker builders. They go, "Gee, I found this great handmade tweeter from Denmark, and it's really the best thing, and how come your guys don't use it?" Well, regardless of the price issues, these people are buying one and measuring it (perhaps) and tweaking a crossover for one particular midrange and tweeter. It's irrelevant to them whether in six months' time the production is the same. That's really an issue that becomes very challenging to professional designers of equipment: maintaining a quality product with all the variations that go on. That's one issue. The other issues are reliability and manufacturability. It really is quite more of a challenge to design a product you are going to build for five years and have it sound just as good as the first one you built. Compared to the sort of obsessive one-off on-the-bench approach, it changes the way you think a lot.

"...sound is subjectively preferable.. .if the maximum stereo separation between the ears is achieved, which is called minimum interaural cross-correlation..."

RANADA: What came after NAD?
KANTOR: I moved back to the States, not wanting to be an expatriate, and I needed work. I began to work for AR as a consultant. Before I had left for England I had been working with AR on a digital signal processor, a very innovative product well ahead of its time. By the time I came back from London that was pretty much in a mess, and management was ill-inclined towards research. They felt that they had spent a lot of money on a project about which they hadn't gotten good advice regarding its marketability. The president of AR and I were talking about shutting down the R & D side of things. And I said, "No, no, you can't do that. Give me a chance to run it. I can produce product. I can recover some of your losses." The first thing I developed was this AR remote control. That was based on some of the digital technology that we had developed for the digital signal processor. And it was proof to the management of the company that if you look carefully at the results of a research project you can get something out of it even if it wasn't what you originally intended. That's the beauty of research. We had these wonderful attenuators that were very accurate and we could make a nice little product out of them. And in fact it sold very well for many years. Based on the remote, I took over the [R & D] department and I went back to management and said, "You know, that was a fun little product, but we're a speaker company. Let's start working on speakers again." And, lo and behold, I happened to have in my back pocket all this work I had done in school that AR had supported. And that became the MGC-1.
RANADA: How successful was that speaker?

KANTOR: It was relatively successful for three reasons. Commercially, it was successful for the price bracket it was in. A couple of hundred pair were sold, which at the time [ca 1985] for a $3600 [a pair] speaker was very respectable. Did it ever make serious money for the company? No, not at all. But that's a different issue. I believe it was successful in establishing the issues I had brought up, *as* issues. Third, I think it was good for the company to regain a little bit of esteem about R & D. It was very successful as a research project. Was it a source of major financial growth for the company? No.

RANADA: What did the MGC-1 teach you that you have been able to use since? Have you used the design principles in other products?
KANTOR: Sure. The work that went into it taught me a lot about two things. One of the things that came up during the development of that speaker [the MGC-1] is quite a part of NHT products till this day. That is, the angle of the radiation with respect to the listener's head: the arrival vectors for minimal interaural cross-correlation. That was perhaps the most obvious benefit.

RANADA: Could you explain that in simpler language?
KANTOR: There are a number of studies that have often been cited about how sound is subjectively preferable—sound reproduction or production—if the maximum stereo separation between the ears is achieved, which is called minimum interaural cross-correlation, a minimum correlation between the signals reaching the left and the right ear.
RANADA: Studies have shown that signals that are like this are preferred to ones that are not?

KANTOR: Right, subjectively. For example, different concert-hall designs yield different amounts of interaural cross-correlation, and it has been shown that the concert hall that produced the minimum interaural cross-correlation tended to be subjectively preferred. The other advantage of minimum interaural cross-correlation is that it gives you a fighting chance at reproducing binaural effects [over loudspeakers]. This has led to a plethora of products, one of the first of which was the Sound Concepts IR-2100, which in fact was a design of mine, as well as Carver's Sonic Holography and Polk's SDA. All these are attempts to reduce interaural cross-correlation by postprocessing. What's less well known is that if you take a single sound source and move it in a circle around the head, there are certain positions of that sound source that naturally yield lower cross-correlation.

RANADA: You mean there are certain angles at which it is minimized?
KANTOR: Exactly.
RANADA: I assume then that the angles wouldn't be the paradigmatic plus/minus 30 degrees.
KANTOR: Exactly.
RANADA: Then what are the angles?
KANTOR: Twenty-one degrees.
RANADA: So you should you have your speakers a bit closer together than the standard equilateral triangle?
KANTOR: Yes. And that's why NHT speakers are designed with 21-degree slants on their face, and that's why the AR MGC-1 was designed similarly—to take advantage of the natural cross-correlation nulls of the human head.
RANADA: So if you wanted to experiment with this you would simply move your speakers inward a bit from the 30 degrees they are probably at now. What would you listen for?
KANTOR: A widening of the ambience.
RANADA: Even though the speakers are closer together?
KANTOR: Yes.

RANADA: What was the other thing you learned from the MGC-1 project?
KANTOR: The other thing was really more of an understanding of what the directivity tradeoffs are, sonically. We all know what the convenience tradeoffs are in regard to directivity—listening position, listening area, room interactions. However, there are also tradeoffs about tonal accuracy involved. It was good for me to sit in a lab and listen to exactly what changed with a speaker having the same axial response but with a variety of directional characteristics. It's one thing to listen to a box speaker and then a dipole, but how do you know what to attribute to what? From a directivity point of view, you don't. There's all kinds of differences in the interference patterns from the separate drivers and the rise times of the systems. You can get lost thinking you know what you're listening to. But, in fact, being able to listen to radiation pattern as a controlled variable was a very good learning experience to me, something that I certainly applied to my later designs.

RANADA: After AR you were a consultant for a while and then you started NHT loudspeakers.
KANTOR: With the [NHT] Model 1 we really tried to take a fresh look at loudspeaker design. We really tried to erase all the preconceptions. We had a couple of goals. One was to rethink the way to build things. One of the most obvious rethinkings that went on was, "Why a square box—what's right about that?" What's right about it is that it's easy to build. But it doesn't put the sound into the room in the correct direction unless you angle the box—and that's inconvenient—and it creates all sorts of internal reflections, comb filter effects, and standing waves that are quite measurable. It's not a mystical thing. You put the same woofer in two different boxes and measure its midrange performance, and you will see the difference. We really wanted to make a speaker that was very inexpensive by today's standards and that really outperformed the status quo in that price range through the application of a number of simplifying principles. One of which was a better [cabinet] shape. Another was a tremendous amount of work in the driver design of the Model 1, much more than you ever see in the mainstream industry. Almost 18 months went into the woofer.

RANADA: Do you make your own drivers?

"...if you take a single sound source and move it in a circle around the head, there are certain positions of that sound source that naturally yield lower cross-correlation."

KANTOR: No, they were being made for us. We followed their production quite closely. We went through many, many iterations in order to achieve not only the performance we wanted inband, but we wanted to eliminate to the largest extent possible the need for a crossover because we felt that was another way to improve the sound and reduce price at the same time. So, in the original Model 1, the only thing the crossover did was to keep the tweeter from falling out. In fact, you could remove the crossover from a Model 1 and as long as you didn't turn the sound up above 80 dB or so, you couldn't even hear the crossover wasn't there. Turn it up louder, and all of the 50 Hz stuff starts getting into the tweeter.
RANADA: So you are one of the few practitioners capable of tackling both electronics and speaker design?
KANTOR: I've done both professionally. I have to admit that I consider myself a loudspeaker designer simply because it's so hard to keep up with both fields, to know all of the drivers that are available, what technologies are coming along, who's manufacturing what. It's such a full-time job that you can't [also] follow who's making the lowest-noise FETs and whose output transistors handle the highest wattage.
RANADA: Keeping up nowadays may mean following driver technologies, but none of the speakers you have done can be said to use any exotic driver technology.
KANTOR: Absolutely.
RANADA: So you don't see a need to go to nondynamic drivers?
KANTOR: Need is a subjective term. Dynamic drivers far and away deliver the most performance for the money, which is a part of what our company [NHT] is about. If I were doing a cost-no-object, performance-maximum system, there are other technologies I might consider; that's a philosophically valid position to take. You just can't beat dynamic drivers for performance per dollar. Dynamic drivers are pretty darn good. In fact, I would go out on a limb and say that dynamic drivers are pretty much unbeatable except for the fact that you need crossovers on them. About the only thing I would want from more esoteric technologies is the elimination of crossovers.
RANADA: And the advantages of things such as electrostatics and other large-panel systems are...?
KANTOR: Two advantages. Large panels have a naturally pleasing psychoacoustical attribute to their radiation pattern. They are very directional on axis and they eliminate early reflections very effectively, yet they fill in the longer time reverberation in the room. If you look at the long-term impulse response in a room from a large panel, it looks quite different from a small box, in terms of there being a large gap between the first arrival and the dense reflections of reverberation, which are much closer to what you get in a concert hall.

RANADA: So there's a longer gap with a flat-panel speaker?
KANTOR: Yes, and also a greater density of the echoes when they do come. And the other advantage the big panels have is that many of them are crossoverless. And being crossoverless, they eliminate a lot of interference-pattern issues.
RANADA: Including the variation of radiation pattern with frequency?
KANTOR: Right, but this is not necessarily such a bad thing as some people think.
RANADA: Once you get past exotic drivers, where do you see progress being made in audio? Let's back up even further and ask the philosophical question. Do you think audio is perfectible? Can the equipment become better than we can hear?
KANTOR: I have now worked professionally in audio for getting on to two decades. I have been privileged to hear many of the best reproduction systems available. That includes both commercially available and laboratory [systems]. I have heard glimpses of perfection. By that I mean, in laboratory settings, recordings of specific things recorded in specific ways that I could not

necessarily distinguish from reality. But those were not systems you could just walk in and put a commercial recording on. I'm forced to say that we aren't that far away from the ability to reliably create perfection in a laboratory setting. But we are very far away from coming close in a commercial environment, from the notion that you can buy a recording and play it back in a way that will recapture what was intended. Now I think there's something that's happening that will force the issue. I think progress will be faster now. For so long the storage media available were so imperfect that it was tempting to concentrate on them as the weak links. The fact of the matter is now it is trivial, for all intents and purposes, to recreate the electrical signal that exists in the recording studio in your house.

RANADA: Even though some high-enders might quibble about that.

KANTOR: The quibbling is irrelevant. We're so close that, even if the high-enders are right, the differences are small and well understood; you could go to 18 bits [resolution] and 50 kHz [sampling rate]—who cares? The point is that we can get such a signal into the house *and it doesn't sound better!* So, in that regard, I agree with the high-enders. I don't agree with their mysticism and I don't agree that vinyl is better than CD. But CD hasn't been a sonic breakthrough; CD has been a convenience breakthrough. Yes, the noise is lower, the top end is less compressed. It is better. But qualitatively, the experience of sitting down in front of a pair of speakers is much the same as it was twenty years ago.

RANADA: Meaning you are not any closer to those glimpses of perfection?

KANTOR: Yes. Many of the annoyances are gone, but those glimpses of perfection are just as accidental and rare as they used to be. In that regard, let's hear it for CD, but nevertheless it [the CD] is pointing a finger at the more fundamental philosophical problem, and that is recording and reproduction methodology.

RANADA: So assuming an arbitrarily complex or psychoacoustically valid method of recording, is it reasonable to expect perfection in a home with commercial pieces of equipment, with all the variability in people's listening rooms?

KANTOR: Well, it's never reasonable to expect perfection.

RANADA: But how about much farther along than we are now?

KANTOR: We can get as far from where we are today as today is from a windup Victrola. I'm convinced of that. It's doable within the next ten years if there is a committed effort.

RANADA: What will that effort involve?

KANTOR: I really don't know. I have my ideas, but I think the important thing to realize is that people are starting to pay attention to the fact that our efforts are presently misguided. That the frontier of audio—and I know this sounds arrogant—is not about more bits or better transistors or vacuum tubes versus solid state. The frontier of audio is about sound fields, what we need in sound fields and what we don't need, something practically nobody thinks about. As a sort of thought experiment, I am sure that if I can create for you a really realistic reproduction of sound through binaural or whatever means, [then] I can [also] stick an equalizer in that chain and knock any frequency band up or down by 10 dB, and you'll identify it but it won't destroy that realism. You might go, "Oh God, that sounds wrong." But it won't make the image, the illusion, collapse.

RANADA: I guess you could apply this reasoning to any of the standard hi-fi parameters—noise, distortion, wow and flutter...

KANTOR: Exactly. And the converse also happens to be true, that perfecting these issues will not yield the lifting of the curtain. People have for so long thought that if we could get that last dB, that last .01, and the curtain will be lifted and the orchestra will be in the room. But it ain't like that.

RANADA: You're actually sounding rather mystical here. Let's get a bit concrete as to what has gone wrong or what the new directions could be.

"So you don't just have two ears. You have ears that are roving 'around the room' and probing the sound field and recalculating it constantly.... they sample that space and...analyze it."

KANTOR: I'm not trying to sound mystical; I'm trying to sound excited. We have to let go of something that has been very precious to us. What we have to let go of is two channels. You have a situation with a room full of energy bouncing around. I don't really care if it's a concert hall, a sound stage, or a playback environment for a synthesizer player. Somehow you have a space of sound, it's three-dimensional, it varies with time, it's amazingly complex. You filter all this down to two signals, and you expect two loudspeakers to rebuild it?

RANADA: The logic has been we have two ears, so why not just two signals?

KANTOR: The simple answer to that: we don't just have two ears. Because our two ears are moving in space, and a lot of the way we hear direction is related to the movement of our head in space. You may be familiar with the psychoacoustical experiments with clamped heads. You clamp somebody's head, and they lose some of their ability to discriminate direction. So you don't just have two ears. You have ears that are roving "around the room" and probing the sound field and recalculating it constantly. These ears exist in a space and they sample that space and they analyze it. Microphones just don't do that and will never do that. So at least in the vicinity of the listener's head you have to create a sound field, an "interactive" sound field, one that the head can move around in.

RANADA: I would assume that the contents of that sound field would have to be exactly what would have surrounded a head in a concert hall.

KANTOR: That's false. That sort of approach will condemn you to failure. If it has to be exact, you'll never be able to do it; the boundary conditions are too complex. How close it has to be we just don't know because nobody has done this analytically; we're just starting to sniff around the issue. We know when it works, which is rare. And we know when it doesn't work, which is common. But we don't know where the dividing line is. We don't know what the specs need to be. In an amp we know what the distortion needs to be, more or less; we know what the frequency response needs to be, to be audibly transparent. Those numbers didn't come from Moses on the mountain; they came from research, years and years of psychoacoustic research. The work hasn't been done yet about how minutely accurate the spatial reproduction of a concert-hall sound field has to be. Who knows? I don't know. The other thing is that the postprocessing of signals is getting good. It's very possible that postprocessing will get so good over the next ten years that it won't be necessary to recreate the sound field; we could simulate the sound field. The hi-fi purists and aficionados cringe at that notion. It seems like artificiality, except that everything we do in this field is artificial—trying to make the air sound like a violin string by moving a paper cone is an artificiality; it's a simulation. You are simulating that vibrating element with a different thing. Reproduction is all about simulation, and I don't think we should be dogmatic about what we will and will not accept. I don't think we should be forced to say it measures right so it sounds right. We have to trust our ears. I have no doubt about that. But once you trust your ears you don't have to be afraid of technology. It's amazing that stereo works at all. If you were to try to understand analytically what a microphone was receiving in a concert hall [when] recording an orchestra, the notion that you'd ever be able to close your eyes and visualize anything even vaguely resembling an orchestra... It's an astonishing, almost serendipitous result.

RANADA: And yet you think we will eventually be able to reliably fool ourselves?

KANTOR: I'm afraid that brings up a different issue: do we want to?

RANADA: But should we be able to create that illusion if it should be our choice?

KANTOR: Right, except that the motivation to fund the research is going to be predicated on commercial application.

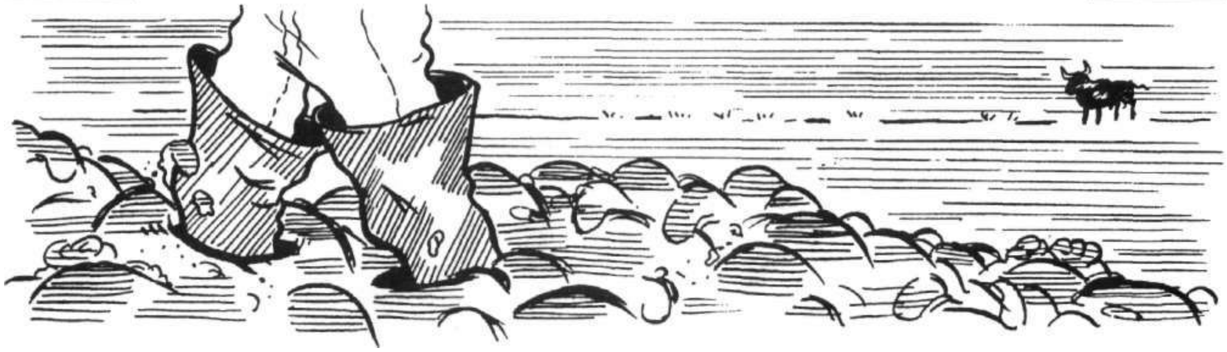
(continued on page 72)

In Your Ear

HI, STEVE. COME ON IN AND I'LL SHOW YOU THE SPEAKERS.
HERE THEY ARE– MY SCHMIGEHGIE SX-100'S!
WOW! THEY'RE BIG!
AREN'T THEY?!
HOW MUCH DID YOU PAY FOR THEM?
I GOT A DEAL. SIXTEEN THOUSAND DOLLARS.
THAT'S STILL A LOT OF MONEY. HOW LONG HAVE YOU HAD THEM NOW?
ABOUT FIVE WEEKS.
SO, JEFF– HOW DO YOU LIKE THEM?
HOW SHOULD I KNOW? NO MAGAZINE HAS REVIEWED THEM YET!
TOM ACZEL

Hip Boots

Wading through the Mire of Misinformation in the Audio Press



***Editor's Note:** David Rich seems to have lost his taste for the blood of incompetent audio journalists because he says this may be his last cameo appearance in this column. (See his byline under the third item.) Well, as George Bush would have said, golly darn! The philosophical/political aspects of "Hip Boots" receive separate treatment in this issue in the "Letters to the Editor" section.*

Stereophile's subliminal response to a zinger.

As the audio community knows by now, *Stereophile* never, never responds in print to our comments on their technical bloopers, no matter how embarrassing, no matter how much their staff owes in intellectual damages to innocent audio enthusiasts. One reason, I daresay, is that we try to make sure that our criticism is unanswerable—but what's wrong with admitting a mistake and correcting it?

Ah, but wait. They hear us all right, even if they pretend not to. In the last issue, this column ended with an editorial parenthesis that said, in part, "I wonder how Larry Archibald, *Stereophile's* owner and President, is able to look himself in the mirror in the morning when he is shaving and tell himself that he is running a credible and responsible publication." A few weeks after that was published, a promotional letter went out from *Stereophile's* advertising department to various advertising prospects. All the letters had the same text, but some were signed by Laura Atkinson, others by Ken Nelson, so I don't know who the author was. The letter said, in part:

"John Atkinson, *Stereophile's* Editor, answers to a 'higher authority': the truth.

"Larry Archibald, our magazine's august proprietor, smiles as he shaves, knowing that *Stereophile* is written to the highest technical and literary standards anywhere in the hi-fi publishing world."

Isn't that delicious? Larry smiles as he shaves! Wonder what made them write that...

As for John Atkinson's "higher authority," some of our readers may not recognize the allusion. It comes from a famous Hebrew National TV commercial, in which Uncle Sam silently mimes as he munches a hot dog, while the voice-over explains that Hebrew National meat products are purer than required by U.S. Government regulations because "we are kosher: we answer to a higher authority"—and Uncle Sam rolls his eyes heavenward. Well, it doesn't really surprise me that John Atkinson's role model is a weenie.

—Ed.

The not-so-secret double life of "AHC."

Anthony H. Cordesman is listed on the masthead of both *The Absolute Sound* and *Audio.* He is a charter member of the high-end audio reviewer coterie, known for many years for his exquisite subjective perceptions of sonic subtleties and meticulously differentiated superiorities/inferiorities that are experimentally unverifiable, i.e., inaudible under controlled listening conditons, i.e., tweako cultist B.S. (or, more charitably viewed, placebo effects). He reviews megabuck power amplifiers with midrange recessiveness/forwardness, preamplifiers with insufficient/exceptional front-to-back depth, the whole tweako canon. That, of course, is exactly what one expects in *TAS* but quite a bit more depressing to longtime readers of *Audio,* where scientific reviewing by engineers used to be the norm. (What do you think C. G. McProud would have thought of the "Auricle" section?)

There's a big difference, however, between Tony Cordesman and your garden-variety tweako audio journalists, the kind we routinely skewer in this column. The latter are nobodies outside the confines of the high-end audio ghetto, whereas AHC is definitely a Somebody. He goes slumming in the audio world only after hours. During the Persian Gulf War you saw him on TV several times a day as one of Washington's top military analysts. At the present time he is military analyst for ABC, professor of national security studies at Georgetown University, author of books on military affairs, and contributor to the Op-Ed section of *The New York Times.* He is obviously a highly intelligent, academically disciplined, responsible professional, necessarily aware of the difference between fact and fiction, between proof and conjecture, between verification and self-indulgent seat-of-the-pants expertizing. That he shows no such awareness in his audio reviews, as distinct from his military analyses, is one of the strangest Jekyll-and-Hyde phenomena in my experience. Is it possible that this professor has never heard of the placebo effect? Is it possible that he has never heard of double-blind listening comparisons at matched levels? Is it possible that he hasn't read any of the numerous ABX test results that have been published? Is it possible that he doesn't know what Ph.D.s in electrical engineering and psychoacoustics think of his kind of audio reviewing?

I could be cynical and speculate that he simply likes expensive toys on extended free loan, so he tells the Mark Levinsons, the Krells, the Wadias exactly what they want to hear in order to keep the toys coming. That would explain the paradox. (Let's face it, the high-end marketeers generally prefer to read about the airy highs and the liquid upper midrange in the reviews, rather than static vs. dynamic distortion and idle channel tones.)

What makes me bring up all this now rather than long ago is the specific case of the Mobile Fidelity Sound Lab "UltrAmp" preamplifier and power amplifier, which Cordesman reviewed in the December 1992 "Auricle" in *Audio,* and which we also reviewed—the preamp in Issue No. 18 (pp. 35-38) and the power amp in this issue (p. 36). We found both units to be clearly inferior equipment (we of the alleged all-amplifiers-are-the-same school!), whereas Cordesman (he of the subtle perceptions, fine distinctions, and lofty high-end standards) writes that both the preamp and the power amp were "capable of good high-end performance." Mind-boggling. His explanatory techie talk in the review is simply regurgitated manufacturer's propaganda, as usual—as an expert on the Marines, he must be familiar with "you talk the talk but do you walk the walk?"—but his soundstaging/imaging/depth/transparency/air palaver is original Cordesman and appears to come from the heart.

I am bewildered. Didn't this veteran high-end guru notice the UltrAmp preamplifier's devastating turnoff transient? Was he totally unaware of the poor channel separation? All right, the preamp is still not a disaster—but that power amp? He writes that "the amplifier becomes more stable under a complex load and can deliver more electrical signal directly to the load." We found that the power amp does just the opposite; for example, into a 4-ohm load it loses close to half its power when the phase angle is ±60°. Its worst-case performance with a complex load is only 9 watts output at 1% distortion! It's a seriously wimpy amplifier, and Cordesman with his fancy speakers and Mark-Levinsonized audiophile sensibility doesn't notice it; he admits the UltrAmp into the high-end club with only mild reservations. But even if he can't hear what's wrong with it, doesn't he feel some sort of minimal obligation to find out from a reliable source what the amplifier is really doing? Why can't he look into the matter a little more thoroughly than some lightweight writing for the Podunk Audio Society's Xeroxed newsletter?

'Tis pity. Tony Cordesman has the intellect, the education, the maturity, the connections, and the enthusiasm to be a truly fine nontechnical, or semitechnical, audio reviewer in the after-hours portion of his double life. Instead, for reasons best known to him alone, he lowers himself to the level of the untutored audio-salon groupies. Come back, Dr. Jekyll.

—Ed.

Once again, Bob Harley gives us the jitters.

A *Stereophile* reader writes to Bob Harley, asking why the D/A processors with the least jitter as measured with the Meitner/Museatex LIM Detector have poor listening ratings (April 1993, page 14). Now, Bob does not answer this question with the correct response—that his listening tests were not done blind or at matched levels, and thus his results are totally random. No, he has a different answer, analogous to how a patent-medicine salesman would respond when a controlled medical experiment shows the cure is worthless. He makes something up and changes the data. He states (April 1993, page 257), "Now, it is intuitive that 100 ps of jitter on a 16.9 MHz clock is a far greater error than 100 ps on a 352.8 kHz clock... .It therefore seems appropriate to express jitter in relation to the clock frequency." He goes on, after a convoluted example of the application of his method, to say, "If you didn't follow all of that, don't worry."

Well, you *should* worry if you *did* follow Harley, since any C student in the second-year digital systems course at Wossamotta U. knows that the jitter in a given clock signal of a synchronous digital system will be invariant with the period of the clock. The jitter in the master clock determines the jitter in all the clocks. For example, if you have a high-speed master clock with, say, 100 ps of jitter and you divided it down (assuming an ideal synchronous digital divider), it has the same 100 ps of jitter.

Now the sampling frequency (fs) of CDs is 44.1 kHz. The complete S/PDIF decoder—here we refer not only to the single chip but also to all auxiliary circuitry, such as multiple PLLs—generates a master clock at 384 fs (16.9 MHz) or 256 fs (11.3 MHz). It also supplies a synchronously divided-down clock at 64 fs (2.8 MHz). The digital filter chip synchronously divides the clock down further, to the final oversampling rate. Irrespective of the system's oversampling rate, the word-clock jitter is determined by the master clock's jitter. (The above statement does not apply to Bob Adams's new Asynchronous Sample Rate Converter chip, but the latter is not yet used in any commercial products.)

But wait a minute, Harley measured 16.9 MHz clocks with an extraordinarily low 6 ps of jitter. How come you cannot just divide the clock down and still have 6 ps of jitter? The answer is that Harley never measured a clock with 6 ps of jitter. This is a most difficult measurement even with very expensive state-of-the-art instrumentation in a precision laboratory setting. Remember that 6 ps is six trillionths of a second! Harley's measurement instrument was giving false readings with the 16.9 MHz signal. In the June 1993 issue, on page 191, he left-handedly reveals that his measurement system had not been taking correct readings: "...the Meitner LIM Detector has been significantly revised. The readings... no longer need to be scaled to the clock frequency." This information is buried in the middle of a review of a digital processor. Nowhere does he state that his original analysis in the April issue was totally wrong.

There are a number of other errors in Harley's original and follow-up article. I could point out each error line by line, explaining why it is incorrect and then supporting my correction with an AES or IEEE reference, but I would much prefer to discuss the whole subject of jitter and its effects on 1-bit and multibit DACs in a separate article. If you want more information now on this complex subject, I recommend AES Preprints 2844, 3416, and 3417 by Steven Harris, AES Preprint 3419 by Malcolm Hawksford, and Masao Sugai's paper in the 1992 International Test Conference (Paper 16.2). Other important references on the subject—by no means a complete list—include AES Preprint 3105 by Robert Gendron, AES Preprint 3361 by Julian Dunn, and AES Preprint 3113 by Nav Sooch and Jeffrey Scott. Harley, by the way, uses a reference in the April 1993 article, but it is not an AES paper! "It has been suggested in discussions I've had with designers that jitter is a much more sonically significant factor in 1-bit than in multibit converters." This is like the patent-medicine salesman producing anecdotal testimony that his cure works, despite what the medical doctors say.

I do not want to come off like the school bully beating up on the weaker kid (even if only academically). *The Audio Critic* has already made the point that Harley is way over his head when he discusses technical issues. So why do we keep bringing it up again and again? Because we think it is important to call your attention to Harley's really major gaffes in order to prevent them from becoming part of your thought process and, eventually, of audiophile folklore. Please note, however, that the real problem here is not Harley but the fact that there is no one on the *Stereophile* staff able to identify the errors in his copy and that outside fact checkers are not used by *Stereophile* to check his copy. I do not want to tell John Atkinson how to run his magazine, but fact checkers are used routinely by other magazines (including *The Audio Critic)* to prevent exactly these kinds of errors. It is time for *Stereophile* to clean up its act and clean up its copy.

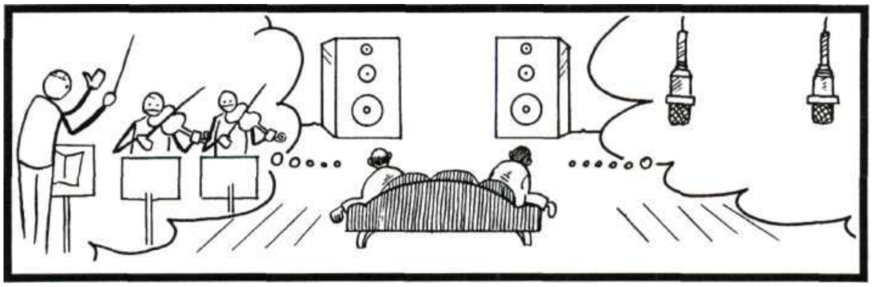
—David Rich

[I have a great idea. Let's pass the hat around to collect enough money to send Bob Harley to E.E. school! Let everyone involved in audio—manufacturers, dealers, distributors, designers, consultants, publishers, reviewers, serious audiophiles, everybody—make a contribution, each according to his ability. I volunteer to be the treasurer of the fund and pledge the first $50 to get it going. The University of New Mexico at Albuquerque is not far from where Bob lives and has a pretty good engineering school; it would be my initial suggestion. What do you say, people?—Ed.]

Major AES Paper by Staff Members of *The Audio Critic*

*Under the sponsorship of **The Audio Critic**, staff members Steven Norsworthy and David Rich will be delivering an important paper titled "Idle Channel Tones and Dithering in Delta-Sigma Modulators" at the 95th Convention of the Audio Engineering Society, Jacob K. Javits Convention Center, New York, NY, Thursday, October 7 (DSP Theory and Applications, Part 1, 1:30 p.m.). The paper discloses hitherto unpublished facts about the performance of $\Delta\Sigma$ converters and presents a simple solution to the principal problem that was found.*

Recorded Music



Editor's Note: *Well, David Ranada has finally come through and is taking over this column—not lock, stock, and barrel, as I and others will be making occasional contributions, but he will be our resident music man from now on. In the next issue I may still do a little catching up on accumulated CDs in my favorably received capsule format, but I expect to phase myself out gradually. High time!*

A Mixed Bag of Recent CD Releases

By David Ranada
Contributing Editor at Large

Introducing myself.

The Editor has generously offered me the dominant portion of the music reviews in *The Audio Critic.* Since you'll be seeing a lot of my music criticism over the next several issues, I think it best that I formally introduce myself in this regard.

Despite my long association with technical writing, first with semiconductors and, for the past 14 years, with high-fidelity electronics, much of my training and education has been that of a musician. My first published works were classical concert reviews for my high school newspaper. In college I majored in music while also taking courses in electronics and computers. My degree is in music, with a specialization in music history (specifically, 19th-century orchestral performance practice and conducting style). I also took a course in Baroque performance practice given by then-flutist Frans Brüggen, who put his students in touch with the then radical ideas of early-music pioneers Nikolaus Harnoncourt and Gustav Leonhardt. I actually got seriously interested in electronics as something more than a hobby through taking a course in electronic music composition, given by Ivan Tcherepnin (son of Alexander). These interests will make themselves felt in my reviews, which for the most part will be of classical recordings, and among these a preponderance of orchestra or small-ensemble works. I will try to avoid unexplained music terminology (as obscure for the outsider as technical jargon often is for musicians), as well as the standard unhelpful music-reviewer cliches (Callas *is* Carmen, Furtwängler's "long line," Horowitz's "touch," etc.).

I am vitally interested in performance-practice issues and especially in tempos and their relationships. Wherever I think it pertinent, the composer's own tempo indications will be discussed. You will sometimes see tempo "measurements" given in beats per minute; this is the standard way of specifying musical speed. I measure performances by timing on a stopwatch the time it takes for ten beats to pass (start on "one," stop on "eleven") and dividing this time into 600, which gives beats per minute (averaged over 10 beats, of course).

Other occasional measurements include CD background noise level. These are derived directly from the CD bit stream, with an Audio Precision System One using its digital signal processing (DSP) feature. Where possible, the readings are taken during "grand pauses," places where the composer stops the sound for a moment. Since grand pauses are often not of sufficient duration to allow a reading of the background noise, I often resort to measuring the short interval of noise at the beginning of a cut or right after the reverb has faded away at its end. The 0 dB level to which all these A-weighted noise measurements refer is full CD output level. Assuming that a disc actually reaches that level, the background noise measurement can then be taken as a measure of dynamic range. (Peak measurements are in the works, at least for selected passages.)

The stated reverb-time measurements are over a frequency range of 400 to 20,000 Hz and are made by watching the rate of fall of the bit stream's rms signal level after a sharp cutoff in the music, either at a final chord or at a grand pause. The value of RT60 (the time it takes the reverb to decrease in level by 60 dB) is usually extrapolated from a much smaller decrease (a fall of 20, 30, or 40 dB). Selecting the two points on the sound decay curve to extrapolate from is a judgment call, so the reverb measurements are meant to be taken only as rough guidelines to the basic sound of a recording, not as definitive statements of architectural fact.

As you can tell, I am vitally interested in how recording quality influences our musical perceptions, and that's why you're also going to see discussion on how the two interact on various recordings. Whenever possible, the first listen to a recording is done following a score of the piece, either from my personal collection or from the immense musical resources of the New York Public Library at Lincoln Center. Score reading puts me on equal footing with both the musicians and the recording producers and engineers, a vantage point from which I can more fairly evaluate the accomplishments of all. Sometimes I will cite measure numbers or cue locations in the score or precise locations on the disc. The latter take the form of [track] min:sec, as in [4] 4:30 (four and a half minutes into Track 4).

One last obsession: orchestral seating plans. You will often see mention of the "old" seating plan versus the "new" one. The old one—established sometime around 1830 and lasting more than a hundred years—places the second violins in the right channel opposite the firsts in the left, and puts the cellos behind the firsts (from left to center). This is the only musically authentic string layout for orchestral music through most of the 19th and early 20th centuries. Most composers we think of as "symphonic" had this seating in mind when they wrote their scores—sometimes strikingly so, as in the cases of Dvorók, Brahms, Tchaikovsky, Mahler, Richard Strauss, and Elgar. It is also a stereophonically more interesting arrangement compared with the modern one, which can be summarized as a crass highs-left/lows-right division (seconds and firsts indistinguishably together on the left channel; cellos, violas, and bases together on the right). Other performance-practice issues related to the stereo image may occasionally be mentioned: Baroque continuo instruments should come from center, for example; or Bartók's "Music for Strings, Percussion and Celesta," which has a specific, and unorthodox, seating plan printed in the score.

Broadway Musical

Richard Rodgers: *The King and I. Julie Andrews (Anna), Ben Kingsley (The King), Hollywood Bowl Orchestra, John Mauceri, conductor.* ***Philips*** *438 007-2 (1992).*

It's not a little infuriating to find a disc of music you love, performed by performers you admire, to be so poorly recorded as to be a caricature of modern recording techniques. Perhaps the source of this disc's problems is it's well-intentioned use of the 1956 film orchestration of the score for a full symphonic ensemble, not an arrangement for Broadway pit orchestra. This in turn has led Michael Gore, the producer, and Joel Moss, the recording engineer, to conjure up the sound of a film-soundtrack symphony orchestra, not a real one, using all the familiar film-sound techniques, none of which are designed to promote sonic realism.

Watching the PBS special on the recording of the album (less a documentary than an hour-long promo video), one could see that the spaciousness of a real concert hall was abandoned for a Hollywood scoring stage, a room with floor space not much bigger than the orchestra itself. Since these venues typically have short reverb times—allowing postproduction control of reverb characteristics thought to be necessary in film sound—recordings made in them are usually souped up with artificial reverberation, and this one is no exception. Everything is reverbed and obviously so.

Everything is also obviously compressed to the extent that the dynamic life has been squeezed out of the music. No symphony orchestra, even one on a soundstage, sounds this compressed. Two of the voices, those of Lea Salonga and the totally unsuitable Peabo Bryson (a casting blunder almost as ridiculous as that of Jose Carreras in Bernstein's Deutsche Grammophon recording of *West Side Story),* are so nontheatrical (in the 1950s sense) that they need amplification and compression to make any impression at all. You'll listen in vain for dynamic inflections out of Miss Andrews or Mr. Kingsley, though their voices are expressive enough.

The whole thing is dreadfully multimiked, and I have the sneaking suspicion that much, if not all, of the voice recording was overdubbed, with the singers not even present when the orchestra was recorded. The PBS show is cleverly edited at places, so that deciding whether this occurred or not is often impossible. What is clear from the show is that Julie Andrews (as Anna) and Ben Kingsley (the King) were in *separate* sound isolation booths performing dialogues and duets! So much for the spirit of interaction vital to a live musical performance and live theater. So much for intermicrophone "leakage" that is so essential in making a multimike recording sound even half natural.

If it weren't for these two formidable performers, the disc would be a total loss. It's already a sonic disaster, like most recordings of musical sound tracks and many original-cast releases—going back to Goddard Lieberson's days at Columbia. Conductor John Mauceri's album notes take great pride in the fact that portions of the (film) score are recorded for the first time since the sound track was made. He deigns to omit, however, the extended but dramatically crucial "Uncle Tom" ballet that was in the film and that has been a highlight of the score since its Broadway premier. It would have easily fit on this disc. You can hear it in okay modern sound on the original-cast album of *Jerome Robbins's Broadway* (RCA/BMG).

A Cappella Vocal

"English Madrigals." Works (18) by Morley, Pilkington, Tomkins, Bateson, Byrd, Framer, Wilbye, and Weelkes. Quink Vocal Ensemble. ***Telarc*** *CD-80328 (1992).*

Elizabethan madrigals are not concert music. Aside from the fact that "concerts" and "recitals" were not even invented until about 100 years after the death of Elizabeth I, madrigals were written to be performed by either skilled amateurs at well-to-do homes or at court. They were a kind of vocal chamber music intended for relatively intimate occasions, often for no audience besides the performers. These simple historical facts account for many of the failings of this disc, which is essentially a madrigal "recital."

First, the recording venue is not a small space with many close-in reflections and a short reverberation time. This naturally would have made for increased intelligibility of the words and their ofttimes complex musical treatment. Instead, for this purely secular music we get the sacred ambience of a church in Groningen, the Netherlands. Measured reverb time is between 2.3 and 2.4 seconds, the same as for many thousand-seat concert halls. In the faster passages (few and too far between, as you will see) this reverb tends to obscure the individual lines, in addition to providing a totally inappropriate churchy mood to the proceedings.

The performers seem to have taken a cue from their surroundings in their overly reverent and lugubrious performances. While the predominant mood of Elizabethan madrigals is indeed somber—the characteristic "fa la la" numbers aside—they certainly were not all be performed at the same tempo as they seem to be here. Single-genre early-music recitals often fall into this trap: of seeming to be in a single tempo, in a single mood, and at a constant emotional temperature. This is musically fatal to a disc playing for 63 minutes.

This sameness of tempo and mood was so striking—and boring—that after I listened through the first time, I went back to measure the tempos for each cut. The basic pulse of the music for five of the 17 tracks fell close to 45 beats per minute. For another six tracks, the tempo was close to 51 beats per minute, perceptibly faster but not sufficiently so to provide enough variety. Half the disc thus hovers around the same, relatively slow pulse (less than one beat per second). The monotony of mood and texture should have been broken up more frequently by the incorporation of other, nonmadrigal works, as it was in a previous Quink disc (Italian madrigals of the same general period on Telarc CD-80209) or by a wider variety of madrigals.

The performers are a skilled group, with very good intonation (pitch accuracy) but without the naturalness of expression that would obtain with native English-speaking singers. On the other hand, Quink's Dutch-accented English is probably as far from Elizabethan pronunciation as the Queen's English would be. (American English would actually lend a more authentic accent to some words.) Inappropriate in ambience though it is, the recording itself is very clean and uncolored, with the voices at a moderate—and unintimate—distance. A-weighted noise was 80 to 83 dB below digital full scale. Despite the good sound quality, you should look elsewhere for a disc of English madrigals. They aren't all this tedious.

"An English Ladymass." Medieval chant and polyphony. Anonymous 4. ***Harmonia Mundi*** *HMU 907080 (1992).*

This immensely popular disc of "13th- and 14th-century chant and polyphony in honor of the Virgin Mary" hit the *Billboard* classical charts shortly after it had received coverage on National Public Radio. And with good reason. The music is, all of it, easy on the ear (provided you have an ear for the polyphony of early music), easy on the attention span (the longest piece lasts seven minutes, most are less than four), and it is extraordinarily well sung, with impeccable intonation and a good deal of expression by the four women who make up Anonymous 4. Given this disc's popularity, it's too bad the liner notes don't include more tutorial information—such as explaining the names used for the individual polyphonic lines: triplum, duplum, tenor, etc.—or discographical information as to where one goes to find more such music.

Of interest to audiophiles is the recording venue: the concert-hall-size acoustically adjustable main scoring studio at LucasArt's Sky walker Ranch. The great advantage of such an isolated controlled space over a more authentic church environment is its lack of background noise, especially at low frequencies. This disc is thus unusually clean and rumble-free compared to other recordings of similar music. Minimalist engineer Peter McGrath has placed his performers at medium distance, so that they are surrounded by a halo of ambience. Presumably the room was operating at a relatively high "setting," since the reverb time measures about 2.25 seconds.

The only reservations I have about the disc are in the tonal quality of the voices—a lack of "presence"—which I attribute either to the mike distance or the frequency response of the mikes used. On a couple of systems I played this CD on, there was a touch of excessive sibilance, though this can also happen naturally in certain reverberant situations. Nonetheless, if you need a female-voice test disc, you can't beat this one.

Orchestral and Choral

Felix Mendelssohn: *Overture & Incidental Music to "A Midsummer Night's Dream" (Suite); Symphony No.4 in A Major, Op.90 ("Italian"). Atlanta Symphony Orchestra, YoelLevi, conductor.* ***Telarc*** *CD-803I8 (1992).*
Johannes Brahms: *Serenade No.I in D Major, Op.II; Variations on a Theme by Haydn, Op.56a. Atlanta Symphony Orchestra, Yoel Levi, conductor.* ***Telarc*** *CD-80349 (1993).*

I was disappointed to find Levi's performances of the Mendelssohn works so clipped, hard-pressed and rushed because this disc's sonics, as well as those of the Brahms disc, are well-nigh perfect for the music. In fact, the distance to the ensemble, the brass/woodwind/string balance and detail, the reverb time (about 2.25 seconds), and the imaging (modern string seating) all add up to some of the best—most musically appropriate—sound quality I have ever heard from any label: these are demonstration-quality discs. Then again, if a record company produces a consistent sound quality, it is bound to eventually come across music for which it is most apropos (romantic-era pieces suitable for classical-size orchestras), even while issuing discs for which it isn't.

The tempos are on the quick side, which is something I usually prefer. But I found the first movement of the "Italian" Symphony impossibly rushed and agitated, something I don't read in the score. Not even revisionist early-music specialists take this movement so quickly: Levi, 9:15; Norrington, 10:25 (on EMI); Mackerras, 10:24 (on Virgin); Harnoncourt, 10:35 (on Teldec). Intensifying the frantic feeling is Levi's continual acceleration through the coda (from Mendelssohn's marking at measure 504 of "piu animato poco a poco"—little by little, more animated/quickly), unlike most conductors who have a pronounced acceleration only over the next few bars and who hold their new, quicker tempo steady. Amazingly, the orchestra—especially the strings—keeps up both here and in the equally frenzied finale. The important exposition repeat in the first movement is taken, as has happened on most recent recordings.

The *Midsummer Night's Dream* music comes off better, with some lovely woodwind and horn playing during the Nocturne and moments of true delicacy in the amazing overture, surely the greatest piece of music ever written by an 17-year old (the rest of the music was composed decades later for a German production of the play). A spirited performance of the famous Wedding March (the postceremony one; Wagner composed the pre-ceremony "Here comes the bride...") is compromised only by tinny-sounding cymbals. All repeats are taken here too.

Those interested in exploring more than the five pieces in the standard suite are urged to track down Kurt Masur's recent performance of the entire score (complete with melodramas and choral/vocal numbers) on Teldec 2292-46323-2, which has a single, unusually versatile actor (Friedhelm Eberle) providing narrative continuity and performing excerpts from the German translation. It's most entertaining for anyone with a knowledge of German. Narratorless, and less complete but musically excellent, performances include those by those by Klemperer (Angel/EMI, sung in English), Kubelik (Deutsche Grammophon, in German) and Maag (London, in English).

Levi's performance of the Brahms Variations on a Theme by Haydn is as pushed and unyielding as his Mendelssohn, with the result that a piece distinguished by its geniality seems merely perpetually agitated. The passacaglia-like finale with its constantly repeated bass line, instead of being the rock-stable goal to which all the moody variations are merely preparation, ends up being as unsettled as the rest of the piece. You may be stimulated but you won't be satisfied.

On the other hand, brisk tempos are all to the good in any performance of Brahms's Op. 11 Serenade, his initial foray into orchestral music written in his mid-twenties. This piece can too easily seem long-winded, for Brahms's skill at musical developmental was still growing and had not yet reached the mastery of dramatic pacing that would power the symphonies and great chamber works. But all the other trademarks of his style are there, especially polyrhythmic devices such as playing duple against triple meters, and a certain heaviness of orchestral sound that takes after Schumann, but without the latter's frustratingly opaque textures.

Aside from being quick, the performance is unsentimental almost to the point of being businesslike: no romantic lingering over particularly juicy phrases here (the deadpan close of the first movement stands out in this regard). But even Levi's fast pace can't relieve the excessive length of the slow movement. This is the most difficult one in the entire score to bring off successfully since, as written, it builds to no satisfactory climax. I've never heard a performance of it that "worked." Brahms would take years to figure out how to write a symphonic slow movement.

The speed of the second scherzo is thrilling, what with all the strings rushing around to keep up. The liner notes have at least one egregious blooper, though, in referring to the trio of this movement as being related to the horn solo in Beethoven's Second Symphony. The whole movement, not just the central trio section, is a rewrite of the scherzo of the "Eroica" (Beethoven's *Third* Symphony). Besides, there is no horn solo in the scherzo of the Second. All repeats throughout the work are taken.

The background noise level of these two discs is very low (-81 to -85 dB). A volume setting higher than that for typical classical CDs (say, from the PolyGram labels: London, Philips, and Deutsche Grammophon) is recommended to faithfully reproduce these discs' very wide dynamic range.

Aaron Copland: *Fanfare for the Common Man; Lincoln Portrait; Canticle of Freedom; An Outdoor Overture.* ***Roy Harris:*** *American Creed; When Johnny Comes Marching Home. James Earl Jones, speaker (Lincoln Portrait); Seattle Symphony & Chorale, Gerard Schwarz, conductor.* ***Delos*** *DE 3140 (1992).*

These are excellent performances, all with excellent sound—except for the big draw, James Earl Jones. His majestic reading of the text of the *Lincoln Portrait* dominates the performance, to the detriment of the music. He is miked so closely, and mixed in so loudly, that the result sounds very artificial, as if he weren't there for the orchestral taping at all. His voice certainly doesn't leak into the orchestral pickup as it should to create a completely integrated acoustical picture. A remix of this cut would blow away its admittedly slim competition.

The overly famous *Fanfare for the Common Man* receives an unusually noble performance (stemming from the steady tempo) that sends the bit stream into clipping just before it ends. The clipping isn't audible, so for once you know precisely how much dynamic range a recording has, since the A-weighted noise level is 84 to 85 dB below clipping.

Interestingly, the interselection noise is "looped" on this disc; a short passage of "room tone" is spliced to itself to make up the necessary intertrack spacing. Also, here Schwarz again uses his accustomed turn-of-the-century split-violin seating plan, which is bordering on the anachronistically old-fashioned for most of the pieces here, which actually sound better with the modern (all violins on the left) seating arrangement.

Sergey Prokofiev: *Alexander Nevsky, Op.78; Lieutenant Kije Suite, Op.60. Janis Taylor, mezzo-soprano; Milwaukee Symphony Orchestra and Chorus, Zdenek Macal, conductor. Koss KC-1016 (1993).*
Sergey ***Prokovief:*** *Alexander Nevsky, Op.78; Scythian Suite (Ala and Lolli), Op.20. Carolyn Watkinson, mezzo-soprano; "Latvija" Choir; Gewandhausorchester Leipzig, Kurt Masur, conductor.* ***Teldec*** *9031-73284-2 (1991).*

The Koss recording is another successful effort by this small but obviously ambitious record company, a pet project of Koss president Michael Koss. Macal's pacing throughout is excellent, as is the sound quality. The only faults in the latter are some persistent low-frequency background noise in the *Lt. Kije Suite* and the overall reticence of the entire percussion section except for the bass drum. In live performance the percussion, unlike the U.S. Mail, always gets through. The choral pickup also is fine, not unlike what Telarc gets on its Robert Shaw records, although the chorus definitely sounds like Midwesteners singing phonetic Russian. Only Janice Taylor's well-sung but matronly-sounding performance of "The Field of the Dead" doesn't quite fit, since the words for this movement are those of a young woman.

Masur's disc is not one of his better efforts. But neither the shortish reverb time (2 seconds) nor the high background-noise level (both pieces are recorded live) is primarily responsible. Instead, it is Masur's urge to rush through the cantata at paces that are almost all faster—sometimes much faster—than Prokofiev's metronomic indications. The "Battle on the Ice" at Masur's tempos imparts no sense of mysterious foreboding at the start and a frenetic feeling to the main battle sequence, with no compensatory sense of the clashing of Russian and German national interests. The biggest plus—aside from the inclusion of the always spectacular-sounding but infrequently recorded *Scythian Suite*—is Masur's chorus, which for once in this piece sounds entirely comfortable singing in Russian. It makes a tremendous difference.

Mozart and Beethoven

Ludwig van Beethoven: *Piano Sonatas No.27 in E Minor, Op. 90; No. 28 in A Major, Op. 101; No.29 in B-flat Major, Op. 106 ("Hammerklavier"). John O'Conor, piano.* ***Telarc*** *CD-80335 (1992).*
Wolfgang Amadeus Mozart: *Piano Concertos No. 20 in D Minor, K. 466; No. 22 in E-flat Major, K. 482. John O 'Conor, piano; Scottish Chamber Orchestra, Sir Charles Mackerras, conductor.* ***Telarc*** *CD-80308 (1991).*

John O'Conor's work for Telarc has always struck me as honest, sincere, pianistically adequate for the music he plays, and—extremely dull. These two discs, superbly recorded as they may be, continue that impression.

He can obviously hit all the notes—the fugue in the "Hammerklavier" is unusually secure-sounding—and consistently chooses appropriate tempos. But there's more to this music than just the right speed, though that *is* half the battle. The other half is expression, which on a piano is obtained by slight deviations from metronomic regularity in the pulse and by phrasing via variations in dynamics.

Even short comparisons with other performances by almost randomly selected pianists (I pulled from my shelves Barenboim for the sonatas and Perahia in the concertos) will show that O'Conor's use of dynamic phrasing is so attenuated as to be practically nonexistent. Everything comes out in a flattened monotone. The music is more interesting than he makes it. And even Mackerras's propulsive conducting of the concertos doesn't seem to have lit a fire under the pianist.

There are alternatives to all these performances, though none of the competition sounds as good. In the sonatas you can start with Brendel (Philips) or Pollini (Deutsche Grammophon) and work your way backward to Schnabel (Angel, mono). In the concertos, you can try Perahia (Sony/CBS) and Ashkenazy (London). And if you're really adventurous, I'd recommend original-instrument performances of the concertos by Bilson (Archiv) and van Immerseel (Channel Classics).

Ludwig van Beethoven: *String Quartet in B-flat Major, Op.18, No.6; String Quartet in F Major, Op.59, No.l ("First Razumovsky"). Cleveland Quartet: William Preucil, violin; Peter Salqff, violin; James Dunham, viola; Paul Katz, cello.* ***Telarc*** *CD-80229 (1991-92).*

The Cleveland Quartet, again of the great American string quartets, is by far the best ensemble regularly recorded by Telarc. So it was with great expectations that I first listened to this disc, the first of a projected complete Beethoven-quartet cycle for the label.

I was not disappointed. In addition to being superbly played and expressively articulated, these are extremely well-paced performances, a fact I attribute to more frequent close approaches to Beethoven's "fast" metronome markings than is usual for a string quartet. Although all of Beethoven's metronomic specifications are controversial, the fact that some movements are musically successful when taken at his quick tempos indicates that the composer's recommendations cannot be dismissed out of hand. This ensemble's tempos suggest that they have considered Beethoven's markings before selecting their own speeds, a most commendable practice.

Here, the first and last movements in general are taken only slightly slower than marked, and the scherzos are taken just below the indicated speed. The slow movement of Op. 18, No.6, which is marked at a rather stodgy 80 sixteenth notes per minute, actually starts out in this performance at a decidedly faster 94 sixteenths per minute (it slows down later). The problem the quartet has with sustaining musical intensity through the weakest performance of the disc, the extraordinary slow movement of Op.59, No.l, stems from its selected tempo, which I felt was too slow even before I measured it (at the start, 74 eighth notes per minute vs. Beethoven's 88). The third movement of the quartet is supposed to be as slow as the first movement is fast, the basic pulsation for both being 88 beats per minute, although I've never heard a performance where the selected tempos, even if wrong, were the same in the two movements. Profundity does not derive from slowness but from saying the right thing at precisely the right time, which cannot happen if the time scale is distorted at a wildly incorrect tempo.

The sound for both quartets (recorded in different halls, though it is difficult to tell this from the sound) is very close-in, but not coarse and raw as could have happened with microphones with a more colored frequency response. The close microphoning leads to the sonic image of second violin being unstable as the instrument makes slight movements in relation to the mikes: sometimes it's to the left of center, sometimes to the right. This type of image shift actually occurs in live performances, but it is compensated for by the audience's ability to see the instrumentalist's movements.

There are also excessive breathing noises captured along with the instruments. While this too is perfectly natural and is in fact an accurate portrayal of what the quartet would sound like if heard from the distance suggested in the recording (6 to 10 feet), the intakes of breath too often signal beforehand what should be typically Beethovenian musical surprises. I hope further releases in the series—which I eagerly await—don't show this mannerism to be this quartet's equivalent of Glenn Gould's humming.

Wagner and Mahler

Richard Wagner: *"Wagner 3" (Siegfried: "Forest Murmurs;" Lohengrin: "Elsa's Dream;" Tristan und Isolde: Prelude, "Brangäne's Warning," Prelude to Act III, "Liebestod;" Die Walküre: "Wotan's Farewell and Magic Fire Music;" A Faust Overture). Allesandra Marc, soprano; Seattle Symphony, Gerard Schwarz, conductor.* ***Delos*** *DE 3120 (1992).*
RichardWagner: *Preludes and Overtures (Rienzi: Overture; Tannhäuser: Overture and Bacchanal ("Venusberg Music"); Die Meistersinger: Prelude to Act I; Lohengrin: Prelude to Ac III; Der Fliegende Holländer: Overture). The MET [Metropolitan Opera] Orchestra, James Levine, conductor.* ***Deutsche Grammophon*** *435 874-2 (1993).*

Gerard Schwarz's third disc of Wagner opera excerpts for Delos is by far the least likable. Despite the fact that he uses the musically desirable and authentic late-19th-century orchestral layout (second violins and violas on the right, first violins and cellos toward the left), the overall sound quality is not particularly transparent. Delos usually records orchestral music a bit too distantly for my taste, but usually the results sounds cleaner than they do here. The unusually scrappy orchestral playing doesn't aid this impression, with the double basses having a particularly difficult time with the very first note of the Faust Overture. A higher-than-usual background noise level also proved distracting, especially the hiss and rumble at the soft beginning of the "Liebestod."

Less satisfactory still is Schwarz's conducting, which is, to put it briefly, dull, with an absence of interpretive ideas except for an impression throughout the Faust Overture that the speed is slowly increasing (the expanding universe at work). He also seems to substitute a tam-tam stroke for Wagner's cymbals at the climax of the Fire Music ([7] 12:34.5).

Worst of all is the vocal contribution of Allesandra Marc, who is too off-mike in the Lohengrin excerpt and much too on-mike during "Brangäne's Warning" (which is supposed to be delivered from practically offstage) and the "Liebestod," obscuring in the latter two excerpts orchestral lines that are always clearly audible in live performances. I can't believe that these are the balances producer Adam Stern and engineer John Eargle preferred; they smack of vocalist intervention. As for her singing, it does not convey any impression that she knows what she is singing about or the dramatic situation her characters are in. *[Even so, that's* some *voice.—Ed.]*

A much better impression of what Wagner's music is about can be obtained from James Levine's disc, one of the best Wagner-excerpt discs in recent memory. He too uses the old string seating, as he has for all his recordings with the Metropolitan Opera Orchestra. He has an excellent feel for how his particular excerpts work in the theater—as attention-getting curtain-raisers that happen also to be great music—and has paced them accordingly, getting grand and satisfying results out of the *Tannhäuser* and *Meistersinger* selections, the highlights of the disc. I do miss the contribution of a seductive female chorus in the Venusberg music, however. Instrumental lines replace them.

Unfortunately, the sound is flawed here too. It is a closely multimiked job in the worst American—and now German—tradition and sounds like a Columbia record from the 1970s. The close miking makes the brass in particular sound harsh, something they definitely are not in live performance, and the strings—lots of strings—are a bit too lush, at the expense of the rest of the orchestra. The measured noise level is around -72 dB, but it drops out to digital zero between selections, probably so you won't hear the contributions of the 8th Avenue subway beneath Manhattan Center, where this recording was made. The noise dropout is audible, and distracting, in a really quiet listening room.

Gustav Mahler: *Das Lied von der Erde. Agnes Baltsa, alto; Klaus König, tenor; London Philharmonic Orchestra, Klaus Tennstedt, conductor.* ***EMI*** *CDC 7 54603 2 (1982-84).*
Richard Wagner: *Die Walküre, Act I. Susan Dunn, soprano (Sieglinde); Klaus König, tenor (Siegmund); Peter Meven, bass (Hunding); Pittsburgh Symphony Orchestra, Lorin Maazel, conductor.* ***Telarc*** *CD-80258 (1990).*

Wagner's scoring has never had it as good as it gets on the Telarc release, which, despite Maazel's use of the modern and inauthentic string seating arrangement, is the best-sounding Wagner on disc, period. Only Haitink's *Ring* cycle (on Angel) comes close among digital-era efforts. Unlike the Delos release reviewed above, this recording places the voices at suitable distances from the microphone: not so close that you hear every swallow while losing orchestral lines, but not so far as to start losing words in the reverb. The brass come through with a low-frequency solidity I've heard before only in live performances, and for once we don't get too many strings (or the right number of strings miked to *sound* like too many strings).

Too bad the performance isn't as interesting as the sound quality. Among the singers, only Dunn sounds halfway involved in the proceedings. König strains to get every note out, which is not dramatically inappropriate at the beginning of the act but certainly is at its rousing, even arousing, conclusion. Maazel is, in contrast, too involved, slowing down recitative passages in an effort to impart some portentous meaning to them and pulling back when Wagner wants the music to push forward (such as at Sieglinde's ecstatic "Du bist der Lenz," [10] 2:28).

A production oddity: König, who should know better, substitutes the word *Sehnens* ("of longing" or "of desire," erotically tinged) for *Sehens* ("of [the sense of] sight") at [2] 4:30, so that what originally had the sense, "My spirits are refreshed, my eye enjoys the blissful pleasure of sight" turns into something like "...the blissful pleasure of lust." Again, not inappropriate, but only at the end of the act, not at its beginning. Someone should have caught this at the original session. *[In Wagner's insufferably turgid, bombastic, pretentious poetry, such a glitch disappears like spit in a swimming pool, and I say this as a passionate devotee of his music, as well as an erstwhile student of German literature.—Ed.]*

It's possible that someone *did* catch the problems with Tennstedt's reading of the Mahler song cycle. Perhaps that's why the recording was released only recently, after having been originally recorded in 1982 and 1984. I had high hopes for this disc, since it is conducted by one of the leading Mahler interpreters of the day. But the sound quality defeats the entire enterprise, being inadequate even for 1982. The music starts out practically in mono, with the violins coming from the center of an image that itself has no sense of space around it. The long last movement has better overall sound quality, but with patches of intermittent distortion(!) and a mood-spoiling overmiked mandolin at [6] 11:53. Other equally important lines are sometimes lost (like the harp at [6] 21:26). That's what you risk with a multimike recording: turning up one mixer slider is like turning down all the others.

König again shows that he has no idea what he is singing about. For her part, Baltsa sounds much more involved, too much so in the reportorial passages in the last movement that Mahler specifically marked "without expression." Tennstedt is not in top form, taking tempos that are at places too fast (second movement) or too slow (first movement). There are many alternatives performance available, nearly all with better sound: Klemperer (Angel), von Karajan (Deutsche Grammophon, midprice), Bernstein (London and Sony/CBS, both midprice), Davis (Philips), and Haitink (Philips, midprice), among others. And don't forget the three commercial Bruno Walter recordings (EMI, London, Sony/CBS) which together form a short history of recorded sound (electrically recorded 78 rpm; mono tape, multimiked (!); stereo tape). •

Coming:

The next installment of the power amplifier survey, with reviews of the top-of-the-line high-power models from Carver, Harman-Kardon, Parasound, and others.

A comparison of very high-priced D/A processors: Deltec Precision Audio PDM Two (with Tl transport), Enlightened Audio Designs DSP-9000 Pro, Krell "Studio." Also, at more affordable prices, a highly original new CD player from Harman Kardon, the Monarchy Audio DT-40A transport, the Cobalt 307 D/A processor, and more.

Our first look at the MiniDisc (MD) perceptual coding system and the latest Sony hardware implementation, with a critical comparison vis-à-vis the DCC alternative.

More loudspeaker reviews, as always, including an exclusive first review (unless someone unexpectedly beats us to it) of the new Win SM-8; plus test reports on the Rush Sound 333 self-powered speaker, ACI "Spirit," Signet SL280, Magneplanar MG-1.5/QR, and NHT's new flagship, the $4000/pair Model 3.3.

Preamplifier survey addenda, including reviews of the remote-control Krell KRC-2, the new Harman Kardon AP2500, and maybe some surprises.

The promised but delayed revisionist analysis of delta-sigma ("1-bit") converters, including a test report on the Crystal Semiconductor DAC that beats the system.

The prematurely announced but definitely in-the-works FM survey (maybe not in the very next issue), plus our regular columns and features (always, in every issue).

Interview with Ken Kantor

(continued from page 60)

It's questionable whether there's enough commercial incentive to get there.

RANADA: I would think that pop music would benefit from being able to place instruments arbitrarily in 3-D sonic space.

KANTOR: Pop music would not only benefit from it, it would be revolutionized. But the ability to place a source through postprocessing and the ability to record a source are related but different technologies. We're getting pretty good with head-related transfer functions that enable you to place sound sources arbitrarily around the listener. But they really don't speak directly to the issue of how to record a natural source and have it come out right.

RANADA: So where are we left here? Are we on the verge of the ultimate solution?

KANTOR: We are on the verge of a conceptual understanding of sound reproduction. Things always get boring before the breakthroughs, because that is what inspires the breakthroughs. Two things have to happen hand in hand. One is that the technology has to improve, but it has to improve through the guidance of the understanding of perception. Psychoacoustics is a field that grew up around issues of telephone and communication. How good does a telephone have to be? How good does a PA system have to be so that the lecturer is intelligible in this big room? What distortion levels will be bothersome? A lot of early work was inspired by these issues. Those were the commercial driving forces. As a result, a lot of the research up to this time has had to do with very fundamental characterizations of our ability to perceive.

RANADA: Fundamental or primitive?

KANTOR: Both—like what are the physiological limits? What are the minimum and maximum sounds we can hear? The highest and lowest pitches, masking, and those very basic conceptual understandings. There is an open book to be filled on our perception of natural sounds and our identification of them, which needs to guide the development of technology. If you were to ask me how many channels I need to make a hi-fi system, I wouldn't know the answer to that.

RANADA: People haven't asked that for a long time.

KANTOR: That's right. Because they've asked it stupidly in the past and they've paid the price for their stupidity.

RANADA: So how does your position as Vice President of Technology and head of research activities for International Jensen enable you to promote this new age?

KANTOR: Like any research facility that's worth anything, we have a combination of long-term and short-term work. Short-term there are whole bunches of problems to be solved. Drivers need to be made better than they are; speakers need to be made better, with more consistency. Long-term we just started a major project in digital signal processing that is very much going to relate to some of these psychoacoustical factors. We are under way in the development of some digital signal-processing equipment to improve sound perception. Where it will lead, I can't say. Will we be the company that will single-handedly pull high fidelity into the future? I hope we tug a little bit, but it's going to take a lot of effort. •

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