Issue No. 15

Spring through Winter 1990-91

The Audio Critic®

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In this issue:

A new and highly accredited contributor to our journal discusses in depth the current state of CD player technology, from DAC architecture and digital filters to analog stage design, and rates the various approaches.

Inspired by the above, your Editor reviews a baker's dozen (count them: 13) CD players and D/A processors.

The introductory first part of our promised exposé of the wire/cable scene makes its delayed appearance.

A short (but not last) installment of the "Seminar 1989" transcript is shoehorned between timelier matters.

Plus our accustomed columns and features, including the return of "Hip Boots" and lots of CD reviews.

The Audio Critic

Issue No. 15 Spring through Winter 1990-91

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

For subscription information and rates, see inside back cover.

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From the Editor/Publisher: More of That Running Commentary

This is the first issue of **The Audio Critic** *in which the editorial "we" has been dropped. I have decided that it's a little stiff and old-fashioned, not right for the 1990's, especially not in the context of a small, informal publication. (Paul Krasner, radical editor, chief Yippie, and weirdo humorist of the 1960's, had an anecdote about an editor of* The New Yorker *who used the editorial "we" in the climactic utterance of his most intimate encounters; since then I haven't been truly comfortable with that time-honored usage.) So, from now on, when the speaker is I, the pronoun shall be the same, and the authors of guest articles will also be encouraged to use the first person singular. Another new step in the same direction is the more liberal use of conversational contractions such as "isn't" and "that's." I somehow started off on the wrong foot and became too rigid a formalist on that subject, beginning with Issue No. 10. Maybe it was a delayed reaction to the studied folksiness of Madison Avenue advertising copy, which had been my bread and butter for so many years. Anyway, what I'm doing in this issue reflects my current preferences in editorial style. I reserve the right to change my mind.*

* * *

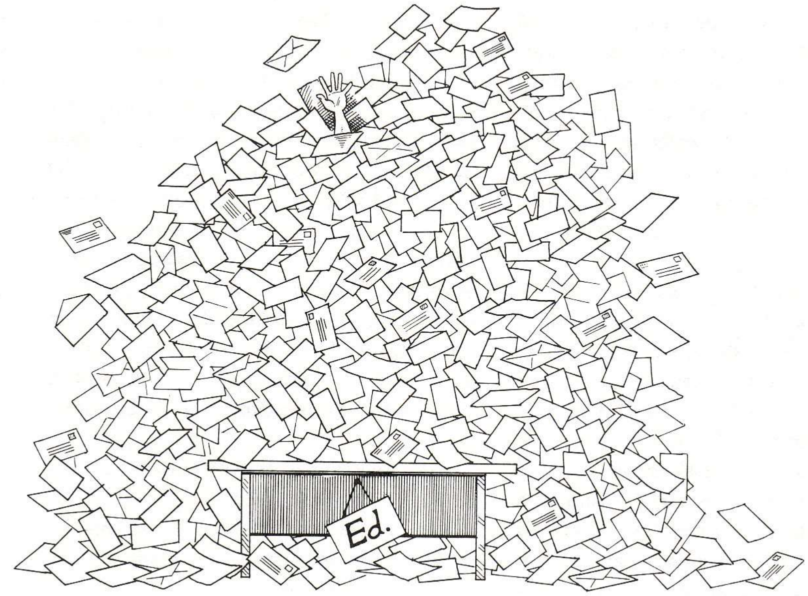
All right, all right, you want to know what happened to the promised quarterly publishing schedule and why this issue is again so late. What happened was that I tried my best but just couldn't get the whole act together for 1990; however, I believe I do have it together for 1991. It's mostly a question of staff, contributors, and ready material well in advance. I expect to have a Spring, Summer, Fall, and Winter issue in 1991. Famous last words, you're thinking, so I'll say no more; when it happens you'll believe it. Meanwhile, look at this issue: it has at least twice the editorial content (as distinct from the paper bulk) of the typical high-end audio journal that comes out more often, and since you're paying $22 per four issues, not per year, you come out way ahead and I get paid less often. That, of course, is my greatest incentive to accelerate. As far as up-to-date audio information is concerned, don't imagine that I'm withholding from you a great deal that I already know because of the laggard publishing schedule. I always try to play catch-up at press time and cram recent findings into the issue about to be printed.

* * *

Let me conclude with a rhetorical question. If audiophile A has read every issue of Stereophile *and* The Absolute Sound *ever published but doesn't know* **The Audio Critic**, *and if audiophile B has read every issue of* **The Audio Critic** *ever published but doesn't know* Stereophile *and* The Absolute Sound, *who has a more solid understanding of the realities of audio, and who is likely to make smarter purchasing decisions, A or B?*

Box 978

Letters to the Editor



Have you noticed that certain high-end audio magazines publish only weak, unconvincing letters of criticism and, on the other hand, always the strongest letters of praise? They obviously want you to think that you are getting a scrupulously fair, balanced sampling of reader opinion. But write them a highly articulate, authoritative put-down that makes them look incompetent or insincere, and they will suppress it. This journal is of a different persuasion. We are not impressed by groupie flattery and we welcome intelligent disagreement, no matter how strong. Try us. Letters printed here may or may not be excerpted at the discretion of the Editor. Ellipsis (...) indicates omission. Address all editorial correspondence to the Editor, The Audio Critic, P.O. Box 978, Quakertown, PA 18951.

The Audio Critic:

I read with great interest your review of Bob Carver's t-mod "Silver Seven" clones. I have seen laboratory data on one pair of Silver Sevens—a pair which may have been defective—but I have neither measured nor auditioned any of the t-mods, so I am not prepared to comment on Mr. Carver's success in duplicating the output of his big tube amp. I would, however, like to address your comments on the wisdom of Carver attempting to replicate the "tube sound."

You state that the output of an audiophile tube amplifier is characterized by high output impedance (over one ohm), a bit of second harmonic distortion, and abrupt rises in distortion at the frequency extremes. These observations are very generally correct but are not universally true. That they should apply at all is a consequence of the design deficiencies in most currently produced tube power amps rather than of the inherent limitations of vacuum-tube circuits themselves.

Most of the performance limitations in tube amplifiers made today are attributable to the single-ended circuitry used in the main gain stage and the plate-coupled topologies used in the output stages, as well as to the output transformers themselves.

Let us first consider the matter of output impedance. If, for instance, a unity-coupled output stage, or a pure cathode follower or a totem pole (series-connected output stage), is employed in conjunction with a correctly designed output transformer, the output impedance can be brought to well below one ohm. I hasten to add that this is not mere theorizing on my part. Unity-coupled amplifiers are currently manufactured by Nestorovic, EAR, and Vacuum State Research, and were formerly made by McIntosh. In all cases, output impedance is or was a fraction of an ohm. A pure cathode-follower output constructed with voltage-regulator-type tubes such as the 6336, 7242, or 6AS7 might practically achieve an output impedance of a quarter of an ohm or even a tenth of an ohm. I can cite no production examples of this type, but a cathode-follower amplifier is available on a special-order basis from Vacuum State Research. A transformer-coupled totem pole output would also yield an extremely low output impedance, though here again a commercial application appears never to have been attempted.

Why have transformer-coupled pure cathode followers or totem poles not been commercially produced? Probably because of their high drive requirements, which simply can't be met with the single-ended voltage amplifying circuitry that has always prevailed in the vacuum-tube realm. With differential circuitry, totem pole or cathode follower outputs are entirely practical, and amplifiers using such output stages do yield low output impedance and good damping. As you're undoubtedly aware, both totem poles and pure cathode followers have been used in commercially produced OTL designs, but that needn't concern us here because OTLs do not have low output impedances.

More important, such unorthodox output stages make possible extremely wideband power delivery without the rising distortion at frequency extremes which you mention. Ordinary plate-coupled tube amplifiers require primary impedance loads in the thousands of ohms and thus critically high turns ratios to achieve good matching to conventional loudspeakers. Even at low power, power bandwidth is inevitably limited by shunt capacitance and leakage inductance in the transformer, but the problems become critical as output approaches and exceeds the 100-watt mark. It is worth noting that until the seventies nobody manufactured a plate-coupled consumer tube amplifier of over 100 watts output. The only consumer tube amps in the 200–300-watt range were the big Macs, which were all unity-coupled (half in the cathode circuit, half in the plate).

With an output stage having a low output impedance before transformation, a transformer can be designed that has extremely low distortion and phase shift in the audio band, and extremely extended frequency response. If, for instance, your primary impedance is 100 ohms, you can get a secondary impedance of 0.1 ohm with a very low turns ratio. Indeed, if the amp is designed with split supplies, an autoformer can be utilized in lieu of a conventional transformer for an almost total elimination of shunt capacitance and leakage inductance. I won't belabor the point. A tube output stage with an inherently low output impedance working into a well-designed transformer will give transistor amps a pretty good run for the money in terms of damping and bandwidth.

I am not implying that those who design plate-coupled amplifiers are fools. Plate coupling makes for simpler, cheaper designs and a much wider choice of output tubes. And it can work extremely well in low-powered units. But today's audiophile wants high power, and I am convinced that plate coupling is not the way to go. And I'm in pretty good company. Frank McIntosh felt the same way.

I might mention in passing that the bandwidth of a plate-coupled tube amplifier can be considerably improved by using a pair of output transformers in parallel, each optimized for one half of the frequency range. It's an old trick, and I believe that Carver uses it on the Silver Seven. Double transformers are also used on the new Cary amps and were used on the now discontinued Meitner tube amps. I don't think the scheme works as well as a pure cathode follower, and it's certainly not cost-effective.

You mention distortion. Not all tubes produce a preponderance of second harmonics. Most triodes do, but pentodes tend to produce a more transistor-like distortion spectrum with plenty of high-order nasties (pentodes were regularly denounced as output devices in 1940's consumer publications). Now, if an all-triode power amp—the preferred design in my opinion—is made fully differential, most of that second harmonic will be bucked out. Operate the gain stages class A and use judicious amounts of feedback, and the small amount of third harmonic will be just about eliminated also. In fact, the active circuitry of a fully differential all-triode tube amp may be designed to produce practically no THD at all, and with values of global feedback many, many dB less than would be required to achieve similar results with a solid-state design. Interestingly, the fully differential all-triode Morikawa amp from Japan has THD in the 0.01% range with a mere 6 dB of global feedback, and most of that distortion is probably due to the output transformer.

Of course, one may object—why bother with tubes when vanishingly low THD values can be achieved with transistors? In other words, what particular merit is there to tube design today?

I think you can gather by now that I still favor vacuum tubes, and that being the case, I wish I could make an irrefutable technical case for their superiority and for the impossibility of a solid-state amp perfectly replicating the transfer function of a properly designed tube amp. I'm afraid I can't settle the matter, but I can state a number of reasons why a tube amplifier should outperform a solid-state type in reproducing music.

The better audio-frequency triodes produce negligible distortion above the third harmonic even at full clip. Bipolar transistors produce a relatively unfavorable harmonic spectrum, especially at clipping, and heavy use of global feedback exacerbates the problem by eliminating low-order distortion while leaving higher-order products. (MOSFETs are a special case; they have a somewhat tube-like distortion spectrum, but total distortion is quite high, and they're hard to drive, especially in multiples.) Class A operation does linearize transistors considerably in this respect but doesn't completely solve the problem, inasmuch as class A transistor amps derate when operated into low-impedance and/or highly reactive loads. (One can, of course, provide the amp with output impedance adjustments, an interesting technique used in the class A solid-state amps made by Sony Esprit and Sphynx, a Dutch manufacturer.) The technique of feedforward also tends to make a transistor output stage emulate the performance of triodes, and may well represent the optimal output topology possible with today's devices, but it still fails to achieve a perfect replication of tube characteristics, and it is very expensive to implement properly and not terribly practical—unless you count quasi-feedforward schemes like Stasis operation, or Quad's "current dumping." In any case, even with feedforward, clipping will still produce lots of high-order distortion, and there are other subtler aspects of transistor behavior which resist the linearizing effects of feedforward, class A operation, differential operation, or any other linearizing technique I know. Transistors have more than a dozen distortion mechanisms—far more than triodes—and these include thermal debiasing, collector/base capacitance, etc. In toto, these are productive of not only high inherent values of THD, but also complex intermodulation distortions which are not adequately represented by simple two-tone tests. Richard Bell, currently of Carillon Technologies, wrote a series of seminal papers on the subject back in the sixties, and his observations still hold good today.

Vacuum tubes are also far better isolated from electrical disturbances than are transistors. They are less disturbed by either RF or reactive loads, and can be operated in amplifying circuits without regulation, something that just isn't possible with transistors. Transistors want a rigidly controlled operating environment, and really superior solid-state amps such as the Threshold Stasis have extraordinary amounts of housekeeping circuitry to keep the signal devices in linear operating modes.

I am not suggesting here that solid-state amplifier design is an exercise in futility or that tube amps are the only valid choice for the music lover. I am only suggesting that the ancient technology of the vacuum tube still has something to offer and can deliver a level of performance that is very hard to emulate.

By the way, if Carver is indeed producing a preponderance of second harmonic in his t-mods, it might interest your readers if you were to explain how he achieves this result. It would certainly interest me. I have seen distortion spectra from literally scores of high-end solid-state amps, and with one exception—a Mark Levinson product—I have yet to see a solid-state amp with a tube-like spectrum. I know that Carver is using neither class A operation nor feedforward, so I am curious.

I must conclude an already overly long letter. Thank you for producing a lively publication and for discussing technical issues generally not covered elsewhere in the consumer press.

Very sincerely,
Daniel Sweeney
Burbank, CA

Yes, as a letter it's overly long, but as a minitutorial on pro-tube philosophy it's about the right length, and I think our readers will appreciate it. Although you have obviously been recruited by the vacuum-tube lobby (not necessarily commercially, just ideologically), your arguments appear to me to range from at least plausible to basically valid. On a few points I and my associates part company with you totally, as future reviews in this journal will undoubtedly evidence. The biggest hole I can punch in your exegesis is that you carefully avoid mentioning the heat generated by tubes, their far from unlimited life span, and their changing performance characteristics as they age. A tube amplifier is not an install-it-and-forget-it type of audio component, whereas a properly designed solid-state amplifier is—and to me that overrides all other considerations. I'm willing to concede that a superior solid-state amp will possibly be more complex than an equally good tube amp, from which it follows that a semipro or dilettante designer/constructor will find the tube approach more congenial.

I think you make too much of the second-harmonic issue. The Carver "Silver Seven" doesn't have such a heavily second-harmonic-dominated signature, or any other peculiarity in its distortion spectrum, that there should be any problem copying its transfer function into a solid-state amplifier. Years ago, Bob Carver sent me at my request a memorandum detailing the successive steps in his t-mod procedure. There are far too many steps, and too many details within each step, to repeat them here. Maybe one day I can prevail on him to write an article. The thing to remember—and what many audiophiles somehow fail to understand—is that you can't t-mod a sow's ear to duplicate a silk purse; in other words, the amplifier being modified has to be potentially as good as, or better than, the amplifier being copied. The basically sound circuit topology and high voltage/current capability of Bob's amplifiers give him a lot of latitude.

—Ed.

The Audio Critic:

I am extremely pleased with the back issues of your publication I received recently. They are well written and cover the finest equipment with an absolute minimum of filler and nonsense.

Your assessments regarding the relative effect of amplifiers, CD players, loudspeakers, interconnects, room and software on sound quality are a breath of fresh air and reason after reading some of the other high-end publications. The others seem to be promoting a mysticism designed to appeal to those with a fear of technology. I am afraid this is a very expensive cult.

I hope to see *The Audio Critic* grow and prosper, but please do not become glossy and flashy like the others.

Please find my check and subscription form enclosed.

Rod Hickerson
Portland, OR

As of the last couple of years, the polarization along the reason/mysticism axis is even worse than you think. I come from an era when audiophiles believed what the professionals told them. Today their heads are filled with garbage fed to them by self-appointed pundits with no credentials, who tell them to distrust the professionals. The witch doctors are elbowing aside the real doctors. Unless a strong countertrend emerges, I fear for the future of audio. This publication alone can't hold the line.

No, we won't become another slick magazine, but we must run color ads on coated stock. It's a matter of survival in a highly competitive situation.

—Ed.

The Audio Critic:

Dear Peter,

Thank you for printing my letter in Issue No. 14 of *The Audio Critic.*

If I may comment on your response to my letter: The point of my letter is and was to get the word out to audiophiles who subscribe to your magazine that all amplifiers do not sound the same! I carry that message to anyone who is interested in getting the best out of his system. I assume that the majority of your readers have that as a goal. I have read every issue of your magazine (journal?) since its inception. I continually get the feeling that your readers are being misled into thinking that all amplifiers sound the same. As a matter of fact, I think that anyone who has been exposed to the ABX debate could be misled. Let me say now that I have no problem with the ABX per se. I do not have any use for double-blind testing. I know that I am intellectually honest and have no motive to use some clue (cue) other than the merit of the component to identify it. I do, however, use blind testing whenever practicable. (Yes, I can guess what component is playing by identifying external cues, but I am honest enough to admit it.)

After reading the roundtable discussion in Issue No. 13, it was clear to me that some individuals could get the impression that all amplifiers sound the same. As a consequence, they could purchase any amplifier they wanted based on factors other than sound quality. Consequently most audiophiles would tend to buy the cheapest amplifier within the appropriate range of specifications, or the one that had the most desirable bells and whistles. This is exactly what a large majority of amplifier manufacturers would want. Then they could concentrate on marketing, distribution and advertising, the things they do best and the tactics that lead to the most consistently high sales figures. I speak with firsthand knowledge on this point. You see, I once believed that all amplifiers were the same, a notion that was reinforced by your magazine, among others. For example, I was talking to a Carver representative a month ago, who told me that all amplifiers sound the same and he could prove it using the ABX comparator. Your magazine's endorsement of ABX double-blind testing, coupled with your claim that any amplifier working within certain parameters sounds the same, perpetuates that misconception. Your response to my letter did little to clear that up. The fact remains that the real-world audiophile (or the nonaudiophile consumer) cannot make the Radio Shack amplifier sound like the Boulder 500 under any real-world conditions. Yet there are many audiophiles who will read your magazine and think that all amplifiers sound the same. Thus they will be easy prey to their desire to save money and buy the cheaper amp. Moreover, they will be tricked by the manufacturer or dealer who would rather sell a lot of cheap amps for a large profit than a few expensive ones for a small profit. The ABX comparison is an intellectual exercise with no real-world application. If the Silver Seven-t sounds exactly like the Silver Seven, then the Silver Seven is a consumer and intellectual fraud. I do not know if you were aware of it when Issue No. 14 went to press, but the Silver Seven is currently being sold as a real-world product through Lyric Hi-Fi (see Bob's interview in the February 1990 *Stereophile*) at the rate of about ten per month.

Having said that, let me respond to your criticism of my letter point by point.

You state that my use of the terms "frankly gorgeous" versus "splitting headache" hardly constitutes expert testimony. Excuse me! You are the expert! That is what I pay you for! I am just a humble audiophile depending on you for advice. If I sound like an amateur, it is because I am. Try this. Hard, brittle, dry high frequencies and overall dry sound versus a euphonic, lush sound. More learned audiophiles attribute this to different orders of harmonic distortions in tube amps than in transistor amps. But then you knew that, didn't you, Peter? (Incidentally, in Issue No. 14, an expert like yourself describes the sound of various components using terms that I would not consider expert terminology—I assume you meant terminology instead of testimony—e.g., page 13, paragraph 2, "...The square-pulse response of the Platinum II is gorgeous..."; page 13, paragraph 3, "...Platinum II's sounded sweet, smooth, open and uncolored..."; page 15, paragraph 2, "...snappy or open..."; page 16, paragraph 7, "...too many flavors..."; page 55, paragraph 2, "...sweeter...")

Peter, those double standards certainly are convenient, aren't they?

As I have already said, I have nothing against the ABX comparator per se. I prefer single-blind testing of the unaltered components, since that is what the real-world audiophile takes home. "Today there exists no halfway respectable opponent of ABX testing who attacks the box itself." I think I can guess who it is you are attacking. I guess you have your own definition of respectable. My definition of respectable is a manufacturer of musical components that represent good value, and the reviewers and dealers who lead me to it. I assure you there are plenty who question not only the ABX box but the motives of those who endorse it. While we are on the subject, I wonder why no one talks about the fact that A/B testing has no real scientific value. The only question is, does the system as a whole do an acceptable job of recreating the illusion of live music at a price you can afford. A/B testing usually is just a gimmick. Do I purchase component A because it comes close to the real thing or because it is better than B? More in line with our discussion here is that the ABX really measures sonic memory. Can you remember what you just heard well enough to match what is now being played and then compare it to something you are now listening to?

I knew you would refuse my offer because amplifiers do sound different, and you know it. You proved it with your audio dollars. The fact is that, even assuming *arguendo* that you could get the Radio Shack amp to sound like the Boulder, the conditions that you were able to do it under would have no real-world application. You are absolutely right, Peter, I did know that you needed an extremely high-quality amp for the lab and, more importantly, for your listening pleasure. (Although the Boulder would not have been my choice.) What do I need in my home to create the illusion of live music? Is your need any greater than mine, or for that matter than that of the thousands of other audiophiles who desperately seek to create the illusion of live music in the home? I think I see it now, Peter: you are not a hypocrite, you are a snob!

What you are really saying is that all things being equal (and you concede that in the amplifier world they never are), you can make two amps sound the same. For that matter, I could compete in the ring with Mike Tyson if you equalized (pun intended) our strength, size, physical skills, knowledge of and experience in boxing, and desire to win. But it just does not happen that way in the real world.

No, you did not say the Radio Shack amp will sound like the Boulder under all conditions, but you implied it. I'm glad you cleared that up. The only way to get the sound of the Boulder 500 is to buy the Boulder 500. That is why you purchased the Boulder instead of any of the cheaper amps available to you, including the Carvers!

I intended this letter to be provocative and to spark debate. Sending it to *Stereophile* and *The Absolute Sound* increased the chances that it would be printed. Indeed, it was printed in *The Absolute Sound.* In answer to your question, I often comment on pending legislation that affects my clients and, yes, I do send copies to all involved.

Thank you for considering my point of view.

Reginald G. Addison
Attorney at Law
Washington, DC

CC: *The Absolute Sound*
Stereophile

Your point of view? You have no point of view, except for a knee-jerk aversion to moderately priced audio equipment and an obvious fondness for the sound of you own voice as you keep repeating yourself. I flatly refuse to defend what I "implied" (in your opinion!), or what "some individuals could get the impression" of, or whatever gave you "the feeling" you elaborate on. The defense would, in any event, consist of a literal quotation of what I wrote, and that you already have in your possession. I said it all then and there. ***The Audio Critic*** *is written for attentive, comprehending readers, not for those in need of a remedial reading class. What I print is what I mean, exactly as you read it, nothing more and nothing less. Your simplistic and subjectively distorted restatement of what I wrote, or what the seminar participants said, is not something I should have to deal with.*

That said, let me comment on some of your rhetorical maneuvers, counselor. You say you are intellectually honest, but I find certain indications to the contrary in the above letter. For example, you say you have read every issue of ***The Audio Critic*** *since its inception, but then why are you unaware that my present position on electronic soundalikes is quite recent, not even fully evident in Issue No. 10? As for your out-of-context quotations of my descriptive adjectives and phrases, maybe a jury of trade-school dropouts from the District of Columbia would fall for such a shabby courtroom trick, but not our readers. Of course I use those basic and very useful words—in conjunction with other words presenting objectively verifiable data. Compare: "Please describe the young woman you saw walking away from the scene of the crime." "Quite frankly, she was gorgeous." As against: "She was a Caucasian brunette, about 24, approximately 5 foot 7, very shapely with long legs, wearing a tight blue dress and little makeup. Quite frankly, she was gorgeous." Yes, counselor, I meant and still mean testimony, credible testimony. The terminology can vary, depending on what is being said.*

You also seem to be innocent of the commercial facts of life in audio. There is more money, for the manufacturer and for the dealer, in selling a $6000 amplifier than there is in ten $600 amplifiers. The entire cost/wholesale/retail structure is far more stringent in the lower price ranges than at the high end, where the figures can be arbitrarily set at whatever level the traffic will bear. As for the Silver Seven "fraud," Bob Carver tells the consumer right up front that the "t-mod" sounds exactly the same, after which the high-end tube freak is on his own, free to part with $17,500 or not. Does that constitute fraud? Should there be a Surgeon General's warning on the chassis to the effect that this product is dangerous to your pocketbook and offers no benefits beyond what is obtainable for $2000? The Silver Seven was originally intended to be a tongue-in-cheek engineering exercise; it was Harry Pearson and Mike Kay who took it seriously and made it into a business.

Your cavalier dismissal of double-blind *testing has me in a quandary. The world's leading authorities in psychophysics consider it to be absolutely essential in order to obtain valid results, not only in audio but in dozens of other areas of investigation; the world's leading medical researchers insist on it for testing new drugs; but Reginald the Lawyer says it has no value. Now I don't know whom to believe.*

All this is more than I really intended to write in response to an inconsequential, hassling letter. Please do not reply again. Be happy with your two soapbox opportunities so far.

—Ed.

The Audio Critic:

Your long efforts to define the perfect loudspeaker give me the feeling that you're trying to prolong the suspense. Let me suggest a concept. A loudspeaker should have the same frequency response in all directions. An equivalent statement is that a loudspeaker should have the same directional pattern at all frequencies. This characteristic is called constant directivity.

As described in *Benchmark Papers in Acoustics,* the loudspeaker was still a recent invention when the best minds in the audio business were called upon to design systems for movie theaters. Obviously, flat frequency response was required over the band to be covered. The designers quickly realized that flat frequency response without constant directivity was inadequate, because there would then be some direction in which the frequency response would not be flat. Floyd Toole rediscovered this just a few years ago. The microphone people have always known that flat frequency response in all directions was the ideal characteristic. Having lots of tricks available, they've been able to come pretty close for a variety of patterns.

For two reasons, it is not natural for a conventional loudspeaker to give flat frequency response with constant directivity. First, a conventional loudspeaker becomes less efficient at high frequencies where the radiation resistance stops rising, so that equalization is required to maintain constant power output. Second, the beam width tends to increase at low frequencies where the diaphragm becomes small relative to the wavelength, so that a horn is typically required to restrict the angular coverage. During the thirties, for reasons of cost, most designers of loudspeakers for the home abandoned correct design forever.

The few people still trying to design constant-directivity loudspeakers have been getting good results for only about twenty years. Most horn systems are still bad enough to lead critical listeners to the conclusion that the concept of constant directivity is wrong. The concept that's wrong, however, is the belief that it's enough for a loudspeaker to be flat on axis.

Sincerely,
William J. Roberts
Toledo, OH

What you say is absolutely correct, but what you forget is that a recording that was balanced by the producer over conventional loudspeakers will in most cases sound far too bright when played through constant-directivity speakers. Unless somebody says, "One, two, three, go!" and the entire audio community changes to constant-directivity speakers at the same time, this problem will remain with us. Furthermore, not all constant-directivity loudspeakers sound the same, either. Yeah, it's a lot easier to design amplifiers.

—Ed.

The Audio Critic:

Although your publication continues at a generally high level of quality, your snide remarks about other publications is inappropriate and insulting to your readers. Just because *The Absolute Sound* has resorted to such a style in the past doesn't mean you should emulate it; in fact, I no longer read that magazine, in part because of that (and I'm sure I'm not alone in this).

The comment about *Stereophile*'s reviews of noncone-type loudspeakers being suspect is wrong, for two reasons. First, the mere fact that Carver is able to modify his ribbon to make it less altitude sensitive means (to me) that this should have been a design parameter from the start; there are enough audiophiles who live at altitudes high enough to incite altitude sickness that they should not be ignored. Second, like nearly all audiophiles, *Stereophile*'s writers prefer various noncone speaker systems, and in general the same ones audiophiles prefer at sea level. If in fact they don't like some systems at their elevation which seem to sound better at sea level, I would not consider that speaker to be a reasonable recommendation. Imagine the audiophile who, shortly after his audition and purchase of Carver's Amazing Loudspeaker at a sea-level dealer, gets it to his home in the mountains and finds himself (rather) disappointed with the sound.

Such attacks are not in keeping with your stated mission and philosophy. Continue to present your data and opinions in ways that readers can enjoy them most, and try to ignore those aspects of your competition which reflect poorly on all of us (the audiophile community—the sensible part).

Yours truly,
Rob Bertrando
Reno, NV

I have two, and only two, reasons for castigating another audio publication. One is that they attacked me, or ***The Audio Critic****, first. The other is that they are spreading major disinformation on the subject of audio.* Stereophile *qualifies on both counts. In 1988, they tried to destroy my credibility via their "Letters" column and did not publish my perfectly civil but highly embarrassing corrective reply. Since then they have been taking potshots at me and my publication at every opportunity. Are you asking me to turn the other cheek? (The first and only persuasive advocate of that policy ended up in a situation I would find unacceptable.) As for disinformation,* Stereophile *is now the most influential disseminator of audio myths, fantasies, and fetishes, the chief apologist for the school of "A blows away B because my exquisite ears say so and I don't have to prove it." To me that constitutes an irresponsible journalistic butt just begging to be kicked. When accountability becomes the rule rather than the exception within the high-end audio community, the prevailing intramural tone will surely be less confrontational.*

I agree with you 100% that the Carver "Amazing Loudspeaker" should not have been released before the altitude sensitivity problem was fixed, as it is now. Bob Carver himself admits it was a serious oversight. That, however, doesn't make Dick Olsher's amateurish and often downright silly loudspeaker reviews more respectable. The man is an undisciplined seat-of-the-pants dabbler in a technology he only partially understands. Yes, a carefully designed loudspeaker should work properly at all inhabitable altitudes, but no one will convince me that electroacoustic measurements made at 7000 feet above sea level are good, reliable practice. I never saw any qualifications or caveats in Stereophile*'s test reports.*

John Atkinson's obfuscatory editorial of May 1990 on this altitude controversy is addressed elsewhere in this issue *(see "Hip Boots").*

—Ed.

The Present State of CD Player Technology: Who Is Doing It Right?

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This is an attempt to clarify at one fell swoop all the diffuse bits and pieces of information that keep cropping up in the audio press on the subject of CD playback and to put that whole technology in a critical engineering perspective.

Editor's Note: Dr. David Rich is the new Contributing Editor on our masthead and a very serious classical music aficionado on top of his formidable electrical engineering credentials. His approach to audio is a little more technical than what our readers are accustomed to; for that reason some of his more esoteric digressions are broken out in sidebars to separate them from the main body of his article. You may want to pass over these on first reading and return to them later—but please, do return.

* * *

Warning: The author will not—repeat, will not—take telephone calls from our readers at the company where he is employed or at the university where he teaches. You can reach him by mail, however, in care of this publication.

* * *

Introduction

This article explores the design of a modern CD player, offering insights into the design trade-offs of midpriced and high-end players. Armed with this knowledge, you will be in a better position to distinguish differences between CD players.

The best source of information about an electronic product is its service manual. Service manuals were consulted extensively in preparing this article. For small American companies that do not publish schematics or service manuals, marketing brochures and interviews with designers were the primary sources of information. Data is summarized in Table 1. We start with an analysis of the most important component of a CD player, the digital-to-analog converter.

Digital-to-Analog Converters

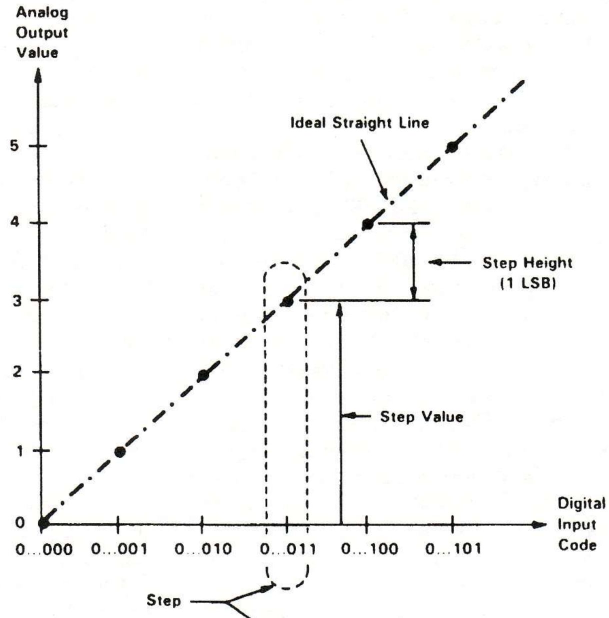
[Burr-Brown 1989], [Tex. Instr. 1989]

The *digital-to-analog converter* (DAC) has the greatest effect on the sound of a CD player. The DAC accepts digitally coded data and produces an analog output in the form of currents and voltages (see Figure 1).

Linearity is the principal performance specfication for a DAC. Linearity is not a new concept in audio. In analog systems, it is the deviation from a linear transfer function, which gives rise to harmonic and intermodulation distortion [Borbely 1989]. The deviation from linearity in analog systems is usually well characterized. For example, bipolar devices have an exponential transfer characteristic. In analog amplifying devices, the distortion increases with increasing signal amplitude. (The crossover distortion in class B output stages is an exception.) The deviation from linearity of the amplifying device is reduced in almost all designs by global feedback. Care must be taken in applying feedback to prevent the formation of dynamic distortion products [Otala 1974].

A DAC's deviation from linearity differs from those characteristic of analog systems. The deviation, generally, is stochastic, randomly varying from one sample of the converter to the next. Systematic linearity errors occurring in each sample of a DAC are correctable by modifying the circuit design or layout of the chip. Distortion worsens as signal level decreases, and feedback cannot be used to linearize the DAC. One researcher has found that very small linearity errors in DACs can "produce audible modulation noise and extremely noticeable audio distortion" [Fielder 1989].

The *step height* of a DAC is the difference at the DAC's output between adjacent steps in the transfer curve of the DAC (see Figure 1). A perfectly linear DAC has steps of equal step height, as shown in Figure 1. Note that the step height has been normalized to 1 unit. This is the smallest analog step at the DAC's output. An *LSB step* occurs when the Least Significant Bit (the last bit of an *n*-bit digital word) is changed while leaving other bits constant. The *resolution* of a DAC is the number of digits necessary to express the total number of steps. For example, a 16-bit DAC has 65,536 steps. There are many definitions of linearity error in a DAC. The most common to characterize the performance of a DAC are *integral linearity* (also called *end point linearity* or *linearity*) and *differential linearity*. Integral linearity is defined as the difference between the actual step value and the nominal step value, as shown in Figure 2. (The actual step values must be corrected for offset and gain errors where absolute DC voltage levels are required to be maintained. These errors are not important in audio applications.) The maximum lin-



CONVERSION CODE

Digital Input Code	0...000	0...001	0...010	0...011	0...100	0...101
Analog Output Value	0	1	2	3	4	5

Figure 1

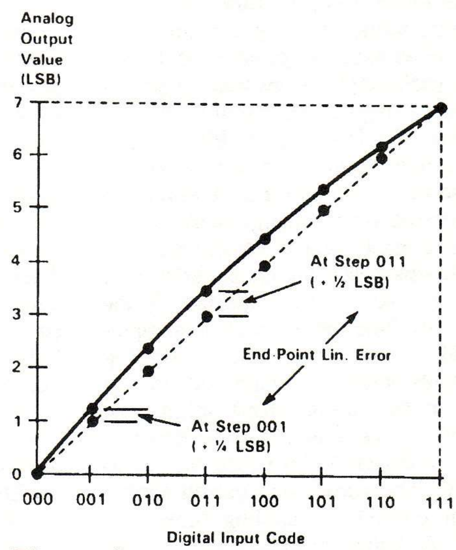


Figure 2

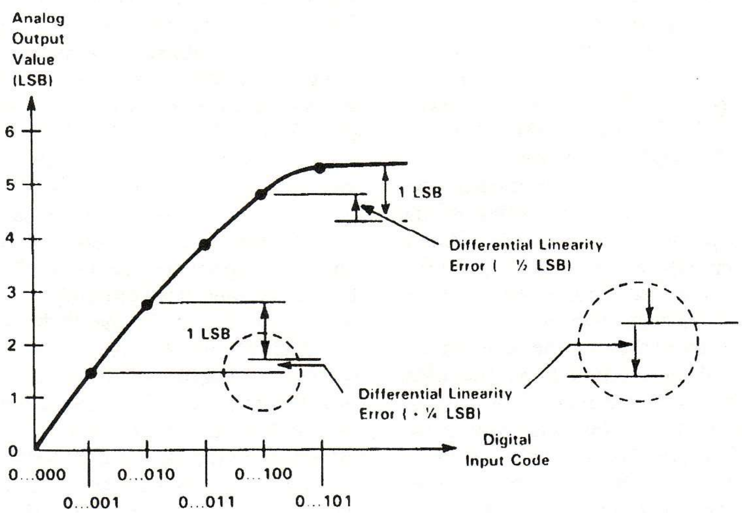


Figure 3

earity error is given in the DAC's data sheet. Linearity errors are often expressed as multiples or submultiples of 1 LSB. Differential linearity is defined as the difference between the actual step height and the ideal value of 1 LSB (see Figure 3). If a DAC has a differential linearity error of greater than 1 LSB, then the transfer function can be nonmonotonic, i.e., the output of the DAC can decrease even when the value of the digital code is increased. A resolution of 20 bits for a DAC is feasible, though the *accuracy* of the DAC is a function of the linearity errors, and the DAC may be accurate to only some smaller number of bits. If the 16-bit DAC has a maximum differential and integral linearity error of ±2 LSB, it is no more accurate than a 15-bit DAC with a maximum differential and integral linearity error of ±1 LSB. In other words, the accuracy of a DAC, not its resolution (the figure quoted by the marketing departments of CD player manufacturers), determines how linearly the signal will be reproduced.

Figure 4 shows the dynamic response of a DAC to a step change in the digital input code. The glitch is a short, undesirable transient in the analog output following a code change at the digital input. The *glitch area*, the time integral of the analog value of the glitch transient, should be as small as possible in DACs used in audio applications that do not incorporate a deglitching circuit. The *settling time* of a DAC (t_{sd} in Figure 4) is the total time required for the analog output to settle within an error band around its final value after a change in the digital input. The error band is usually ±1 LSB wide. However, settling characteristics to wider error bands are important if the DAC is to function without a sample-and-hold circuit. The value of settling time varies with the magnitude of the change in the digital word value. The *conversion period* is the time between successive digital codes being applied to the DAC. The conversion period should equal or exceed the settling time. The number of words presented to the DAC in one second is called the *word rate*. The word rate is a reciprocal of the conversion period.

The Best DACs

Only the Burr-Brown DAC729KH digital-to-analog converter has a guaranteed 16-bit differential linearity error of one bit (it can be externally adjusted to 1/4 LSB) and an integral linearity error of 1/2 LSB for 16-bit resolution. The DAC729 can also settle within 1/2 of a 16-bit LSB when it is sampled at a 4× interpolation rate. Unfortunately, the DAC-729, which is not designed for consumer audio, sells for $197 (in quantities of 100). It is not a single monolithic chip. Rather, it is a hybrid circuit incorporating numerous state-of-the-art custom-designed chips. The price of the DAC729, a product of hybrid technology, is 10 times greater than the maximum price a DAC in a high-end CD player would cost.

The $12,000 Stax DAC-X1t digital-to-analog processor uses a similar hybrid circuit, the DAC D20400 manufactured by UltraAnalog. The UltraAnalog circuit guarantees 18-bit differential linearity, but no specifications are supplied for integral linearity or settling time to a 16-bit LSB. The differential linearity specification holds only at room temperature. The Stax unit uses tubes in its output stage; thus the DACs may not perform to specs because of elevated operating temperatures. The distortion specifications of the Stax unit show that the very low distortion at the output of the UltraAnalog DAC is compromised by the tube output circuit's nonlinearities. *[Ha-ha!—Ed.]*

Although typical specifications for differential and integral linearity are given for a consumer DAC, these ratings are not guaranteed. Instead, a set of THD values is specified. In this test, a sequence of digital words which represent sine waves of different amplitudes is transmitted to the DAC, and THD at the output is measured. Quoting from [Burr-Brown 1989], "THD is the measurement of the magnitude and distribution of linearity error, differential linearity error, noise and quantization error. [Distortion, attributable to quantization error, can be eliminated if a dither is added to the sine wave {Lipshitz and Vanderkooy 1988}.] There is a correlation between the THD and the square root of the sum of the squares of the linearity errors at each digital word of interest. However, this does not mean that the worst-case linearity error of the D/A is directly correlated to the THD." The DAC with the lowest guaranteed THD levels is the UltraAnalog DAC D20400 hybrid. Since the THD of a DAC can be difficult to measure because of the low absolute value of the distortion products, an alternate, simpler test is often used to assess DAC linearity. This test is called *gain linearity*. Gain linearity is the measurement of the deviation of the amplitude of the sine wave's fundamental component from the ideal value, for sine wave signals of varying amplitude ranging from full scale to below an LSB. This is commonly called the linearity test, and the errors are reported in LSBs. This test, widely used by audio magazine reviewers, should not be confused with the more stringent integral and differential linearity tests. Philips researchers have developed a test signal which explores differential linearity over a wide dynamic range [Dijkmans and Naus 1989]. The test uses a 400 Hz sine wave recorded at –80 dB in combination with a .03 Hz sine wave at –20dB. This test will expose differential linearity errors that are not found using single-tone THD measurements.

If a DAC does not have a low glitch energy, or if it does not settle within a small percentage of the sampling period, it must be followed by a *sample-and-hold* circuit. The sample-and-hold samples an analog input signal value and then holds the instantaneous input value upon the command of a digital control signal. Figure 5a is an idealized sample-and-hold circuit. In the sample mode, the switch opens and the capacitor stores the value of the input voltage at the point the switch opens. The circuit samples the output of the DAC after the converter has nearly settled to its final value. This value is then held on the capacitor when the next data word is presented to the DAC. Figure 5b shows a simplified circuit diagram of a sample-and-hold. In the sample mode, the circuit acts as a unity-gain inverting amplifier. In the hold mode, the capacitor holds the value of the output at the time the switch is opened. As will be discussed, the implementation of this particular circuit has its problems. Ultimately, it is not possible to build a cost-effective sample-and-hold that does not distort the input signal.

All of the current DACs designed for use in high-end CD players operate without a sample-and-hold circuit. These include the Philips TDA1541A (16-bit resolution), the Burr-Brown PCM56P (16-bit resolution), the Burr-Brown PCM58P and PCM61P (18-bit resolution), the Analog Devices AD1856 (16-bit resolution) and AD1860 (18-bit resolution). The PCM56P was revised so it can be used without a sample-and-hold. Older CD players that used this chip had a sample-and-hold circuit. (The high-priced Burr-Brown DAC729KH, unfortunately, does require a sample-and-hold circuit. The UltraAnalog DAC D20400 includes a sample-and-hold as part of the hybrid circuit. The differential linearity and THD specs for the D20400 include the sample-and-hold circuit.) The precursor to the PCM58P, the PCM64P, was the first 18-bit resolution DAC. It required a sample-and-hold circuit for proper operation. I compared the sound of the Pioneer PD-91 and Sony CDP-707ESD, which used the PCM64, to that of their respective successors, the Pioneer PD-71 and Sony CDP-X7ESD. The latter are similar but not identical to the older models, with the principal difference that they incorporate the PCM58P. Although my listening comparison was not double-blind or even single-blind,

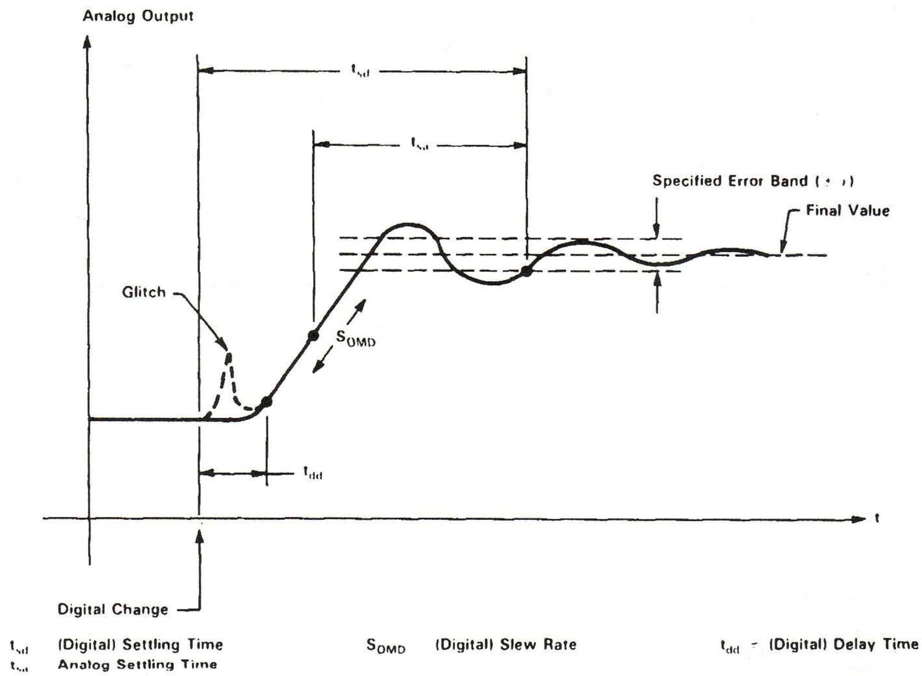


Figure 4

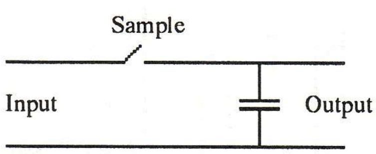


Figure 5a

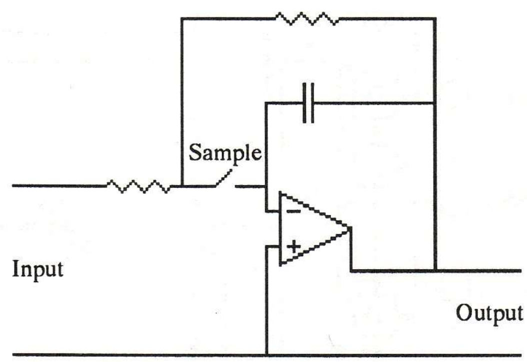


Figure 5b

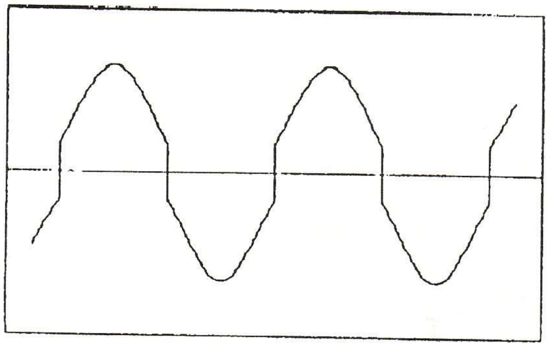


Figure 6a

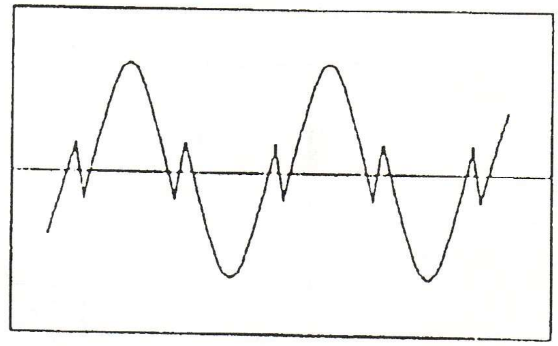


Figure 6b

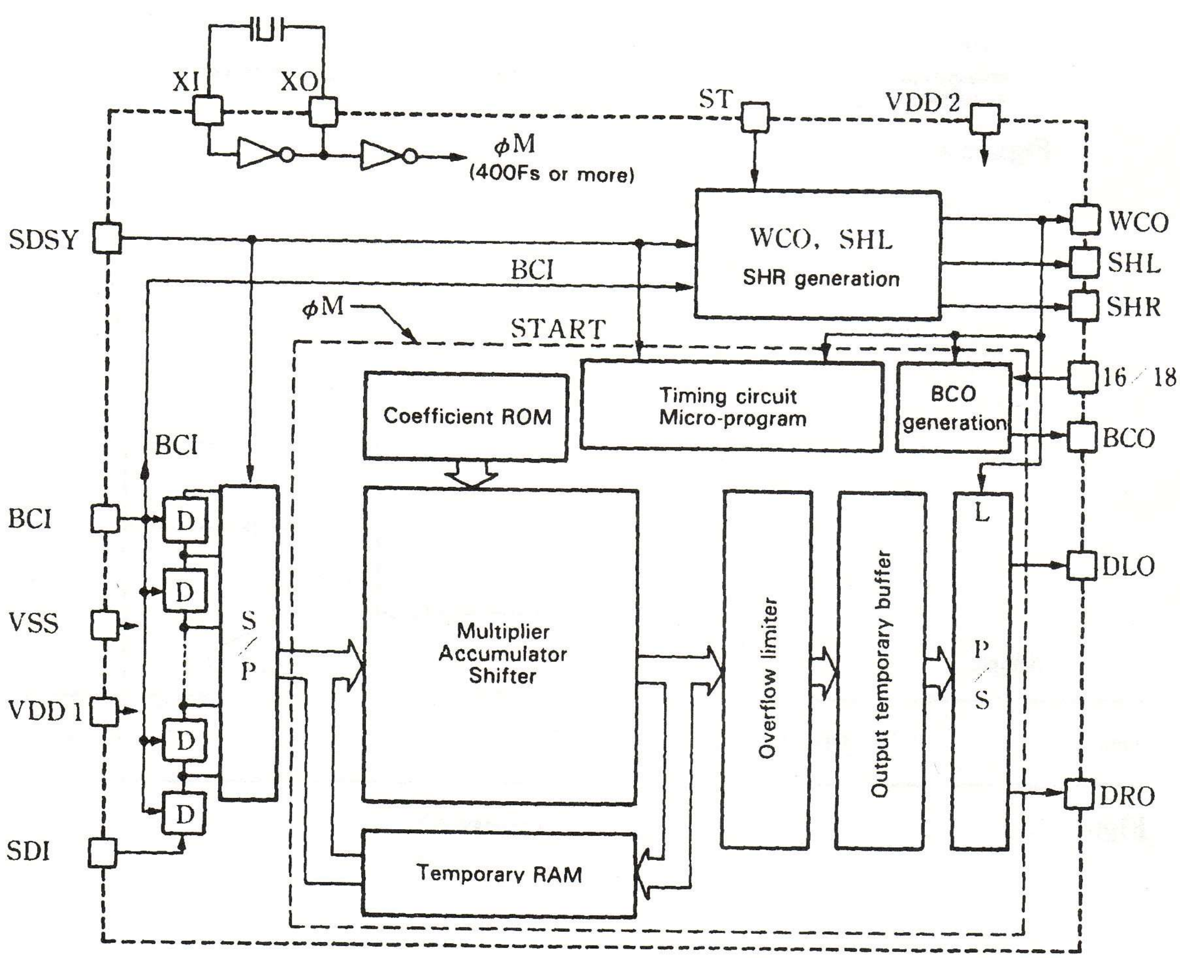


Figure 7

nor at exactly matched levels, my impression was that the sound of the new units is significantly more open and less "electronic." I attribute this difference to the elimination of the sample-and-hold stage.

One potential problem in removing the sample-and-hold circuit is the feedthrough of digital clock signals to the analog output of the converter. Digital signals typically have a peak-to-peak amplitude of 5 V. These signals enter the DAC to encode the next digital word into the DAC and they determine when the conversion process begins. Normally, these signals would not be running during the period that the output of the DAC is sampled. When the DAC's output is continuously connected to the input of the analog circuit, a change in a digital signal's value may slightly affect the analog output. This is probably a third-order effect, though some designers of high-end CD players minimize it by controlling the rate of rise and fall of the digital signals connected to the DAC. They also align the transition time of all the digital signals that enter the DAC.

Denon, JVC, Technics, and Yamaha add external circuitry to increase the resolution of digital-to-analog conversion two bits beyond the resolution of the DACs they are using. High glitch energy is one problem with these systems. Consequently, a sample-and-hold circuit is required. The Yamaha CDX-1120 eliminates the sample-and-hold with a new low-glitch implementation of their floating-bit DAC. In addition, the problem of matching the components added to the DACs in these systems tends to limit the accuracy of these CD players. These systems, I believe, are incapable of matching the accuracy of the best available monolithic DACs. I would greatly hesitate to purchase a CD player with external circuitry for the purpose of enhancing DAC resolution.

Denon (in the DAP-2500 digital audio preamplifier), Kinergetics (in the KCD-40), and Yamaha (in the CDX-1120) use a novel approach for reducing the nonlinearity of a DAC, in which a pair of DACs are wired in a push-pull configuration. One DAC in the pair receives digital information which has been modified so that the polarity of the signal entering the DAC will be inverted. The analog current-to-voltage converter takes the difference between the respective current outputs of the two DACs. Matched even-order nonlinearities that appear at the output of the two DACs are then cancelled. This approach is successfully adopted in analog circuits characterized by closely matched even-order nonlinearities. Most DAC nonlinearities, however, are caused by random processes, and they do not match between dies. Thus, only the small systematic nonlinearities will be canceled. This approach is very expensive because each channel is serviced by a pair of DACs. A single, highly linear DAC will outperform two less linear DACs wired in a push-pull configuration. Hence, the push-pull topology should be used only with the highest-grade DACs available.

It is not possible to fabricate the internal circuit components in a DAC to match closely enough to achieve 16-bit accuracy. Burr-Brown and Analog Devices adopt a technique called *laser trimming*. A laser adjusts the value of the critical resistors on the chip during the initial testing of the silicon wafer. After this test, the wafer is split into individual chips, which are then placed in plastic packages. The packaging of a die can effect its performance through exposure to mechanical stress. Final packaged parts are retested. Not all of these parts perform equally. Devices with the best THD performance are separated. These DACs are then sold at different price points, depending on their respective THD performances. DACs are generally classified into one of three grades. Suffixes are attached to the part number to indicate its quality.

Ranking the DACs

A current ranking of DAC accuracy, in ascending order of improved performance, is as follows.

Category 1:	AD1856	AD1860
	PCM56P	PCM61P
Category 2:	AD1856-J	AD1860-J
	PCM56P-J	PCM61P-J
Category 3:	PCM58P	
Category 4:	AD1856-K	AD1860-K
	PCM56P-K	
Category 5:	PCM58P-J	
Category 6:	PCM61P-K	
Category 7:	PCM58P-K	

The PCM58P and PCM61P are guaranteed to meet their distortion performance specifications without a deglitcher. In my opinion, only the DACs in the last four categories should be used in a midpriced or high-end CD player. The PCM56P-K and AD1856-K, albeit 16-bit DACs, offer better linearity than some of the 18-bit resolution devices. These 16-bit DACs are cheaper than the less accurate 18-bit models, though marketing considerations curtail their use. One of the leading DAC designers calls the extra two bits "marketing bits." For a CD player with a four-figure price tag, I would consider only a PCM58P-K or a PCM61P-K. Engineers at Madrigal, Theta, and CAL have selected the PCM61P-K device for their machine, claiming that it offers better sonic performance. Theta has performed measurements which they claim show the PCM61P-K to be more linear than the PCM58P-K; however, they are now using the AD1860-K. Some manufacturers use lower selection grades of DACs, asserting that the selection process is done in-house. This is hardly convincing, since the binning process ensures that the lowest-grade DACs will have the poorest performance; all DACs with superior performance have already been removed. (An exception to this rule occurs if the number of top-grade DACs produced exceeds the demand for them. When this happens, some of the top-grade DACs are marked with a lower grade).

The Burr-Brown PCM1700P and Analog Devices AD1864 contain two DACs on one silicon chip. This allows the use of a single chip for stereo applications. The Analog Devices part comes in blank and J grades, and its performance is identical to that of other Analog Devices parts of the same grade. The PCM1700P yields performance that is slightly poorer relative to the PCM58P for a given part grade. (The comparisons are complicated by the fact that the THD levels of the PCM58P are specified at a different sampling rate than those of the PCM1700P.)

The architecture of a practical DAC causes the worst differential linearity error to occur at the *most significant bit* (MSB). The MSB is the largest incremental output change obtained by switching a single input bit. This is unfortunate because the MSB change occurs when the output of the DAC passes through zero. For small-amplitude sine waves, the differential linearity of the MSB can have a significant effect on signal distortion. Figure 6a shows a sine wave when it is reproduced by a DAC with large positive differential linearity error at the MSB. Figure 6b shows a DAC with a large negative linearity error. The DAC of Figure 6b is not monotonic. To reduce distortion, the Analog Devices and Burr-Brown DACs allow an external trim adjustment that trims the differential linearity error at the MSB change close to zero. The PCM58P allows adjustment of bits two through four in addition to the MSB. Designers debate whether these adjustments offer additional sonic performance improvements. The accurate adjustment of these potentiometers is difficult in a production environment, and independent laboratory tests confirm that many units are shipped with the adjustments incorrectly performed [Lipshitz and Vanderkooy 1988]. *[See also the CD playback equipment reviews in this issue.—Ed.]*

The Philips TDA1541 uses a different architecture than the Burr-Brown and the Analog Devices DACs. The architecture applies a proprietary technique called *Dynamic Element Matching*. The architecture allows the DAC to achieve 16-bit linearity without laser trimming or external adjustments. The elimination of an external adjustment is especially advantageous since a trim pot can change with age. The top-of-the-line TDA1541A S1 offers first-class performance, but it is not possible to rank this DAC with the American DACs above because the guaranteed specifications for the American DACs and the Philips are different. (The American manufacturers guarantee distortion and Philips guarantees differential nonlinearity.) In the CD players tested for review in this issue, the best DACs from Burr-Brown were found to have lower distortion at –90 dB than the Philips TDA1541A S1. Until the new American DACs that did not require a sample-and-hold circuit made their appearance, the TDA1541A S1 was used almost exclusively in all high-end CD players. Manufacturers such as Sony, CAL, and Kinergetics have now chosen alternative chips from Burr-Brown and Analog Devices. Moreover, newer companies entering the field (e.g., Krell, PS Audio, Aragon, and Proceed) use the Burr-Brown devices. Philips remains the preference of European companies.

Burr-Brown and Analog Devices are continuing the development of audio DACs. For this reason, you should make sure that any very expensive CD player or decoder you are contemplating to purchase can be upgraded to the newer DACs when they become available. The most recent DACs from Burr-Brown and Analog Devices are the PCM63P and AD1862 respectively. The PCM63P uses a new topology which steps away from zero in small steps in both directions to reduce low-level nonlinearity. The AD1862 uses a digital offset technique which shifts the zero level away from the MSB transition. The PCM63P-K and AD1862N-J chips have better linearity performance than the PCM58P-K. The PCM63P-K data sheet lists slightly better THD specifications at the –20 dB and –60 dB levels in comparison with the AD1862N-J. (Again, the comparisons are complicated by the fact that the THD levels of the PCM63P are specified at a different sampling rate than those of the AD1862. In addition, the THD for the AD1862N-J is an A-weighted measurement.) The lower grades of these DACs are not as linear as the PCM58P-K. The THD level of the UltraAnalog DAC D20400 hybrid is 6 dB lower than that of the PCM63P-K at –90 dB. The resolution of the new DACs is 20 bits. (The higher resolution also satisfies the marketing department.) It takes approximately 6 to 12 months after the introduction of a component before it begins to appear in a commercial product. Pioneer is the first company to announce the use of the PCM63P. The new Pioneer CD players are the PD-73 and PD-93. These units are expected to become available by the time this article is in print. *[That's sufficient lead time.—Ed.]*

Digital Filters and Interpolators

[Lipshitz and Vanderkooy 1988]

All quality CD players now place a digital filter and interpolator (the term *oversampling* should be reserved for an A/D converter sampling at a rate much faster than the Nyquist rate) ahead of the DAC. A digital filter affords significant reductions in the complexity of the analog filter. With a 4× interpolation rate, the analog stage could be formed with only two active gain stages. With an 8× interpolation rate, a single active gain stage can be used. A well-designed digital filter should introduce virtually no signal distortion. This is in contrast to a high-order analog filter, an analog circuit which, owing to unavoidable nonlinearities in the active devices, can introduce

Inside the Digital Filter

The topology in which a digital filter is implemented is a highly specialized microcontroller called a *digital signal processor* (DSP). A block diagram of a digital filter, the Yamaha YM3434, is shown in Figure 7. The circuit blocks which constitute the DSP "engine" are all inside the smaller dashed rectangle. Other blocks within this rectangle (BCO generation, P/S, output temporary buffer) and those outside it are specialized digital circuits for formatting data from the CD player's disc-reading circuitry. These circuits also make the data available to the filter's DSP engine and send the output of the latter to the DAC. As for the DSP engine itself, it functions as follows: The coefficient ROM stores the digital words which control the filter's shape. The *multiplier/accumulator* performs the arithmetic operations required for the filter. The accumulator stores partial and complete computations from the multiplier. The *shifter* manipulates the digital words during the multiplication process. The temporary RAM block is required to store the output of the accumulator because the processing of the cascaded filter blocks is performed in parallel, and the data emerging from the accumulator is not the data for the next computation. The ROM, RAM, and the arithmetic unit are controlled by the timing circuit block. The *microprogram*, which is stored on a ROM internal to the timing circuits, controls the operation of the filter.

The word length of the coefficient ROM partially determines the accuracy of the filter response. The effect on passband response is not important, e.g., the NPC SM5805, which has a short 16-bit word, is flat ±0.00025 dB. The added word size has a more important effect on the stopband rejection. The Sony CXD1144BP, which has 293 taps and a 22-bit coefficient word length, has a stopband rejection exceeding 120 dB. When two digital words are multiplied, the resultant word length at the output of the multiplier is the sum of the input word lengths. This word is too large to use and must be shortened (requantized). The process of shortening introduces quantization distortion [Dijkmans 1989]. Some digital filters truncate the word. This is a less desirable process than a rounding operation. Lipshitz observes that, in addition to rounding, a dither signal must be added during the multiplication process to ensure that all quantization artifacts are removed. No current monolithic DSP chips use dither. (Dither has been used in certain Theta and Wadia digital decoders, but given the constantly changing filter algorithms used by these companies, it is unclear if dither is used in current production models.) Note that adding dither at the input of the DAC has no advantage, provided analog dither was added during the recording process. Adding dither is standard practice in modern recordings [Lipshitz and Vanderkooy 1988].

The bus of data connected from the arithmetic unit and the temporary memory is called the data path. The word length of this path is another parameter which affects the filter's performance. The data path word length is usually the word length of the DAC, though it may be larger if noise shaping is performed. The marketing departments have recently taken notice of the word length of the coefficients, accumulator output, and data path. They are using these in advertising copy, perhaps with hopes that readers will confuse these larger numbers with the resolution of the DAC.

The optional noise shaper can round the data at the accumulator. Normally, the noise power is constant from DC to half the sampling frequency. A noise shaping filter is an *IIR filter*, a filter with an infinite impulse response, which redistributes the quantization noise shape. A noise shaper reduces the noise power in the audio band and increases it outside the audio band. The signal-to-noise ratio of the signal bandlimited in the audio band increases. Noise shapers exhibit low-level instabilities called *limit cycle oscillations*. Proper rounding operations and use of dither prevent this, as significant distortion and frequency-response variations. An engineering trade-off must be made between the reduction in the complexity of the analog stage and the increase in nonlinearities at the DAC output. This is because the linearity of the DAC can be degraded when the conversion period of the DAC is reduced. A DAC operated at an 8× interpolation rate will have half the conversion period of one operated at a 4× rate.

Digital filters often use a set of cascaded, finite impulse response (FIR) filters. The sampling rate of each filter section is increased relative to the section which precedes it by a power of 2. The bulk of the filtering takes place in the first section, since this section operates at the slowest clock rate and is therefore easier to design. Finite impulse response filters, which are difficult to design in the continuous time (analog) domain, have the significant advantage of being linear-phase if the coefficients are chosen properly. Because of the linear phase characteristic, such a filter exhibits better time domain response to a pulse than an analog filter.

The smoothness of the passband, the slope of the transition band, and the attenuation of the stopband are determined by the size of the FIR filter. The size is measured by the number of delay blocks in the filter and is called the *tap length* of the FIR filter. (There are N+1 taps in a filter with N delay blocks.) The comparison of the tap sizes of different digital filters is valid only if the filters have the same interpolation rate. A properly designed filter's passband flatness is, remarkably, less than ±0.0001 dB. This is insignificant compared with the frequency response errors in the analog chain.

A designer of CD players who wishes either to improve upon off-the-shelf filter chip designs or to incorporate additional ideas of his/her own will need plenty of money. A design team of engineers would require at least a year, owing to the chip's complexity, to develop and fabricate such a chip, which may need upwards of 100,000 transistors. The cost of engineering time and materials would be in the $200,000 to $500,000 range. This investment is well out of the reach of small American audio companies. Because a custom chip, fabricated for a single company, would be used in much smaller quantities than a standard product, the cost per chip would be much higher than that of off-the-shelf digital filter chips.

The only solution for a small American audio company designing a state-of-the-art product is to design with a general purpose DSP chip, external RAM, ROM, and a number of smaller *glue* chips. The glue chips, as the name implies, form the interfaces between the other chips in the system. Because of the large computational requirements, more than one DSP chip may be required (one for each channel, for example). The cost of this group of chips is much larger than that of a single monolithic device. The Motorola DSP56000 is proving to be the most effective chip for this application. The high prices of the Theta ($2000 to $4500), the Krell ($3500 to $8950), and the Wadia ($1995 to $7995) reflect the cost of implementing the digital filter with general-purpose DSP chips. Some of these units run at much higher interpolation rates (16× to 64×) than the monolithic filters. The manufacturers of these units claim that the monolithic chips do not perform the interpolation function accurately. In the monolithic chips, an ideal brick-wall filter, which is required by the sampling theorem for the exact reconstruction of the input data, is approximated by the FIR filter. A brick-wall filter has a sin x/x impulse response. The time domain form of the sampling theorem states that when a sin x/x function is convolved with samples of a bandlimited in-

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apparently does dither added in the recording process. Noise shaping is used on two 18-bit digital filter chips, the NPC SM5803 and Sony CXD1244. The noise shaping can be turned on and off under software control. Therefore, a service manual may not show whether a given CD player is using noise shaping.

Interpolating digital filters are also plagued with potential overload. This overload arises because signal amplitude at the output of the filter can be greater than that allowed by the word length of the filter. The amplitude increases because of the *Gibbs phenomenon* [McGillen and Cooper 1974], which occurs when a signal is bandlimited and all its Fourier coefficients are not present. An example of the Gibbs phenomenon is seen in test reports on CD players as an oscillation on the top and bottom portions of a square wave. The problem is worsened by an increased filter interpolation rate. Lipshitz calculates that two bits of headroom are required in a 4× interpolating filter. Attenuation of the input signal to the digital filter will solve the problem, but attenuating the input penalizes the signal-to-quantization-noise ratio of the filter beyond acceptable levels. Hence, most monolithic filters detect the presence of an overload and allow the filter to clip. It is unlikely that the filter will clip in the presence of music signals as distinct from test tones. The extra bits available from an 18-bit DAC could provide the headroom, though this is not done on current monolithic filters. The designers of these filters prefer to use the extra bits to reduce the quantization noise introduced in the truncation process at the output of the accumulator.

Currently, there are only four manufacturers of monolithic digital filter chips: NPC, Sony, Yamaha, and Philips. The prices of the chips are dependent on the complexity of the DSP section. A filter with more taps, longer coefficient ROM words, longer data path, or a larger accumulator will be costlier.

The Sony CXD1144 18 by 8 filter is the most complex chip to date, and it is priced at double the competing NPC SM5813 (the similar SM5803 adds a noise shaper and other features that do not impact on the performance of the DSP core) and Yamaha YM3414. Sony's newest design, the CXD1244, has not as yet been adopted by any manufacturer other than Sony. The performance of each of the 8× interpolating filters is summarized in Table 3. Sony has not quoted the multiplier size, coefficient word length, or filter tap length in its data sheet of the new CXD1244. The ripple rejection of the CXD1244 and its stopband rejection are slightly inferior to those of the CXD1144, indicating a simplified design relative to the CXD1144. The lower cost of the CXD1244 also supports this notion. The CXD1144 is often cited by circuit designers as the best-sounding single-chip digital filter. It should be noted that none of the differences in passband or stopband characteristics given in Table 3 should be audible. Therefore, it is unclear how the increased complexity of the CXD1144 results in better sonic performance. Most Philips TDA1541A's are used in tandem with the Philips SAA7220P/B digital filter, although some Sony digital filters have functional modes which make them compatible with the TDA1541A at a 4× interpolation rate.

The CXD1244 chip allows the reproduction of very small signals without switching the MSB bit because it can apply a small DC offset to the digital code emerging from the filter. This feature is unique to the CXD1244. Very small signals are reproduced with lower levels of distortion. The DC offset would cause the positive peak of large-amplitude signals to exceed the maximum digital word size of the filter, thereby clipping the positive peak of the signal. To avoid this, all signals entering the digital filter are slightly attenuated in amplitude.

The Philips digital filters have a very similar feature but differ from the Sony CXD1244 inasmuch as the DC offset cannot be defeated.

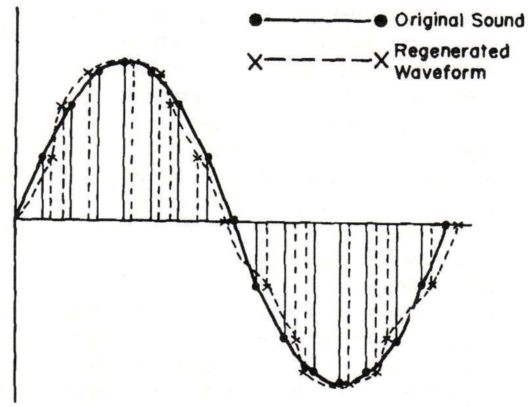

Figure 8

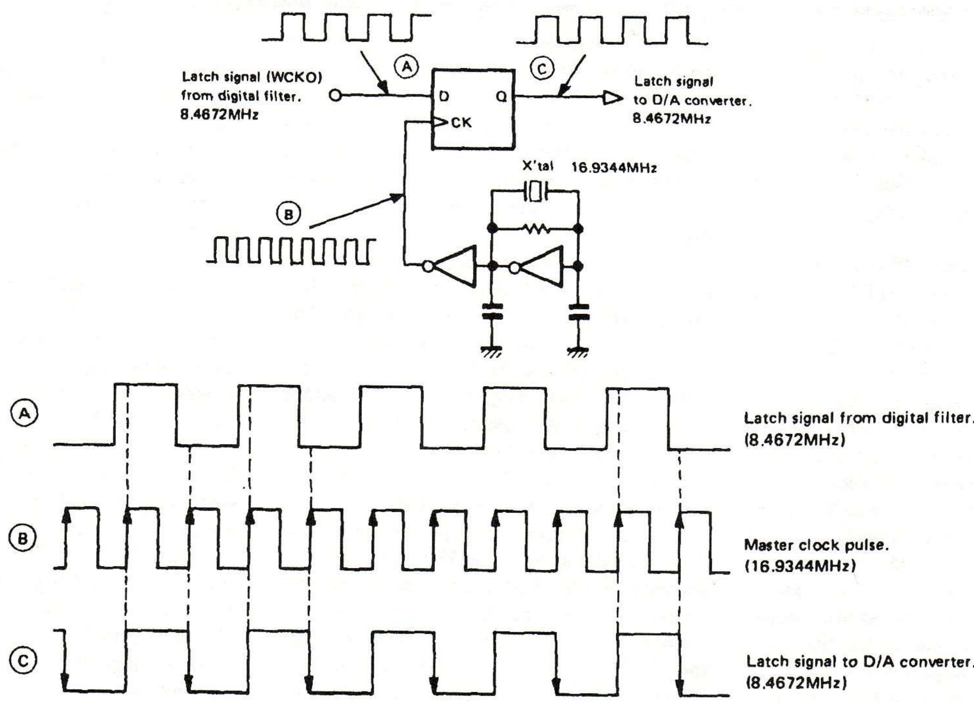

Figure 9

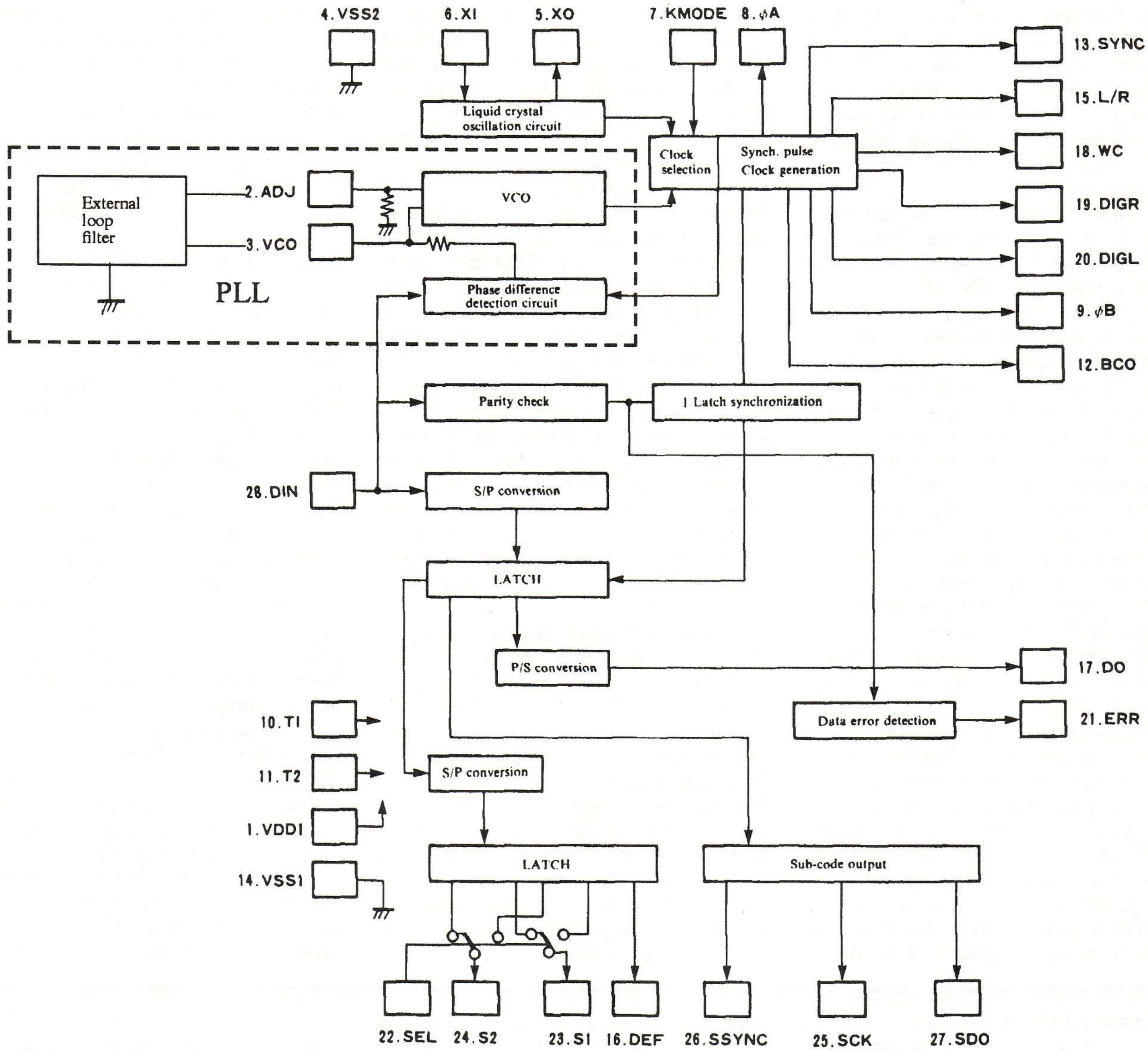

Figure 10

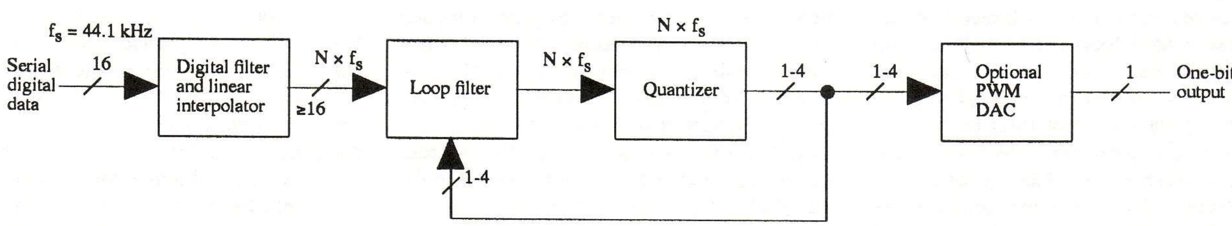

Figure 11

put signal, the bandlimited input signal will be reconstructed exactly [Papoulis 1980]. The impulse response of the FIR filter is finite and of the sin *x/x* function is infinite. The coefficients in the filter are slightly modified to account for this [Papoulis 1980]. Coefficient modification routines are well-known and give good results. Wadia, Krell, and Theta claim that the method used by the monolithic chip manufacturers is less than ideal. In the case of the Theta, the coefficients are also adapted, depending on signal conditions. The Wadia and the Krell perform the interpolation function directly in the time domain rather than the frequency domain. As none of these companies have published their current algorithms in the open literature, it is impossible to assess their methods. Wadia has published information on an earlier time-domain algorithm using Lagrangian interpolation [Moses 1987]. The performance criterion for a digital filter that performs interpolation and smoothing is the *Mean Square Estimation Error (MSEE)* [Papoulis 1984]. None of the manufacturers have published any data showing that their filters have a lower MSEE than a low-cost monolithic filter. *[Touché!—Ed.]*

Time-domain interpolation algorithms have the major disadvantage of not rejecting the out-of-band spurious signals with as great attenuation as a FIR filter [Cezanne 1988]. Martin Colloms has found this problem in his measurements of the Wadia 1000 and the Krell units [Colloms 1989], [Colloms 1990]. The Wadia sales literature points out that the digital impulse response of the Wadia system rings less than a standard digital brick-wall filter. The amount of ringing in the impulse response is directly related by the Fourier transform to the stopband rejection of the filter. The lack of ringing in the tails of the digital impulse response curve for the Wadia system is a direct result of the poor stopband rejection. The sampling theorem requires that the signal at the input of an analog-to-digital converter must be bandlimited to one half the sampling frequency. An analog or digital brick-wall filter must be included in the digital tape recorder to satisfy the sampling theorem. Thus, the impulse response of a complete digital audio system (analog in to analog out) will be that of a brick-wall filter regardless of the response of the playback system to a digitally generated unit impulse. *[Touché again.—Ed.]* Martin Colloms also found that the frequency response of the Wadia 1000 and the Krell units was down by 3dB at 20 kHz. This indicates that the Wadia and Krell algorithms have not been optimized for maximum passband flatness.

In a recent piece of promotional literature, Wadia implies that the results of the sampling theorem cannot be applied to music signals. They argue that in the derivation of the sampling theorem the Fourier series is used. They claim the Fourier series cannot be used to represent the stochastic music signals. This statement is completely false. It has been shown that the sampling theorem is equally valid for bandlimited random signals [Papoulis 1984]. One hopes that the Wadia copywriters misunderstood the Wadia engineers.

Wadia has implemented the glue circuitry in a programmable gate array manufactured by Xilinx. The interconnection of the circuitry is controlled by software programmed into a programmable ROM (PROM). The circuitry for the SPDIF decoder (see below) is also implemented in the programmable gate array. Wadia can thus correct errors or upgrade the circuit configuration of the decoder box by changing the code in the PROM.

Phase Jitter and SPDIF

A sampled data system's performance is critically dependent on the accuracy of the sampling time interval. Variation in the absolute timing of successive spacings is called *phase jitter* (or *jitter*). A sampled sine wave signal corrupted with phase jitter will, when examined on a spectrum analyzer, appear as a narrow Gaussian-shaped band of signals. The resulting signal spectrum is very similar to that of an analog signal reproduced from a turntable or tape deck with flutter. The effect of jitter on the time-domain plot of a sine wave is shown in Figure 8. The crystal oscillator which generates the clock signal is generally jitter-free. Time-base jitter can arise if the power supply signals to the crystal oscillator become noisy. Kenwood proposes that clock noise induced by the CD tracking system gives rise to time-base jitter. According to Kenwood, this explains the claim that CD rings and disc stabilizers change the sound of a CD.

The obvious solution is to ensure that the supply to the crystal oscillator is well-regulated. A problem is that the crystal oscillator circuit is often incorpo-

Inside the SPDIF Circuitry

A block diagram of the industry standard YM3623B SPDIF receiver chip is shown in Figure 10. The SPDIF signal enters the DIN pin. The various digital data and clock signals recovered from the SPDIF signal appear on the output pins of the chip. An analog circuit, a *phase-locked loop* (highlighted in Figure 10), implements this clock recovery. The clock recovered by the phase-locked loop (PLL) will contain more jitter than the crystal-oscillator-generated clock signal in the CD player.

One major source of jitter is the *voltage-controlled oscillator* (VCO) in the PLL. The amount of phase jitter in the VCO output signal is dependent on the VCO's circuit topology. The phase jitter of the VCO can appear at the clock output of the SPDIF decoder [Gardner 1979]. The type of phase detector (marked phase difference detection circuit in Figure 10) used in the PLL also strongly impacts upon the phase jitter at the output [Fourre 1989]. It is often not possible to optimize the loop filter of the PLL to acquire the SPDIF signal quickly and also produce a clock that is low in jitter. Generating a stable clock signal in the outboard box (where the quality of the clock signal is important) and using an additional signal line to send the clock to the CD player—this clock replacing the internal clock of the CD player—would be a better solution for connecting an outboard decoder box to a CD player. Sony's former top-of-the-line CDP-R1 and DAS-R1 combination adopts this approach. This is surprising, given that Sony established the SPDIF format. Unfortunately (for the rest of us), all other CD players generate only an SPDIF output and will not accept an external clock signal.

The clock generated from the VCO cannot be resynced with a clean clock generated from a second crystal oscillator in the decoder box (as was done in the JVC, Kenwood, and Sony CD players). The two crystals, one in the CD player and the other in the decoder box, run at slightly different frequencies. Resyncing is possible only if the two signals have the same frequency, but they may still have different phases. To equilibrate the frequencies, an elastic store which accepts data at one rate and reclocks it at another would be required. If the CD player runs faster than the decoder box, then data accumulates in the elastic store, since data enters at a faster rate than it leaves. If the CD player runs slower than the decoder box, the elastic store fills its memory with data before placing data on its output line. Data could be read out at the faster clock rate from the data in the elastic store. As the CD plays, the amount of data in the elastic store's memory decreases as data is removed faster than it is replaced. A large, uneconomical elastic store would be required because of the abundance of data on a CD and the variation in crystal frequencies in a CD player. (Technics has very recently announced that they are working on a practical implementation of rated in the digital filter IC. Noise on internal chip power supply lines cannot be removed. The solution to this problem is to build the crystal oscillator as a separate circuit and drive the digital filter with the output of this circuit. This approach is used by Stax, Sony, and CAL, among others. Another source of time-base jitter occurs when the crystal oscillator signal is divided down to the word rate of the DAC in the digital filter chip. The source of the jitter is again power supply noise. Resyncing (realigning) the edges of the clock waveform at the output of the digital filter with the master clock will reduce this timing jitter mechanism. Sony, in its CDP-508/608/X7ESD players, performs the resyncing with a custom IC, the CXD8003. Kenwood uses a similar chip called DPAC (Digital Pulse Axis Control) in its DP-8010 player. The basic DPAC circuit is shown in Figure 9. The system in Figure 9 allows the latch signal to the DAC to change only on the rising edge of the master clock. The practical implementation of the DPAC circuit is significantly more complex than the circuit shown in Figure 9.

JVC has found [JVC 1989] that additional sources of jitter include noise coupled into the clock line from adjacent signal lines by mutual inductance and mutual capacitance. In addition, signal reflections in the interconnection lines between ICs can cause jitter. The K2 Interface is a functional block developed by JVC to suppress jitter. The K2 Interface combines optocouplers and a data resyncing circuit. The K2 Interface is placed before the digital filter; some jitter may be reintroduced by the filter IC. I think these manufacturers may be attacking a second-order problem while leaving other major design flaws in their players unresolved. For example, the Sony CDP-508ESD uses an unselected Burr-Brown PCM58P DAC. It would have been preferable to eliminate the CXD8003 from the 508 and channel the cost savings towards an upgraded DAC.

The problem of jitter is more significant when the SPDIF *(Sony-Philips Digital Interface Format)* signal is employed. This is the data format used in connecting a CD player to an external digital decoder box [Rumsey 1989]. The SPDIF uses a single cable to transmit data recovered from the CD player. The data is specially encoded so that the clock signal can be recovered from the data. (See sidebar for technical details.)

Some subjective reviewers have cited differences in the sound of the output of a decoder box when driven with different transports. If the differences are real—and that remains to be proved with properly conducted ABX tests—then jitter is the culprit. Some high-end manufacturers propose expensive transports with low jitter. This is of little consequence, since the jitter problem can only be completely eliminated in the decoder box. Wadia and Theta claim that the bandwidth of the optical cable system used for the SPDIF format is inadequate. This can give rise to jitter. Coaxial cable designed to transmit wide-bandwidth digital signals is recommended instead. Unlike audio cables, these coaxial cables will transmit the SPDIF signals precisely. An example of such coaxial cable is RG-59, which sells for 20 cents per foot. You can bet that the snake-oil manufacturers will be introducing SPDIF interconnects at $100 apiece.

One final observation. I found the PS Audio "Digital Link," which uses only the Yamaha YM3623B chip, to have an excellent sound quality when used with a high-quality transport such as the one in the Philips CD-80 or the Pioneer PD-71. This indicates that the jitter level of the YM3623B is sufficiently low so that it does not significantly affect the sound quality of modern decoder boxes that use it. The early Philips and Sony decoder boxes had less sophisticated SPDIF decoders and may have more jitter. The Yamaha CX-1000U preamplifier/decoder incorporates several clever circuits placed around their YM3623B chip to reduce jitter even further. Yamaha is trying to keep these circuits for themselves. They are not discussed in the YM3623B chip data sheet. New SPDIF decoder chips are currently being introduced by Philips (SAA7274P) and Crystal Semiconductor (8411). I have not received data sheets on these chips and thus cannot comment on their performance relative to the Yamaha.

"Bitstream" D/A Conversion

The ability to integrate highly complex digital systems using low-cost CMOS IC technology is one of the important advances in this decade. Early in the development of fine-line CMOS technology, research focused on more economical implementations of ADCs and DACs. Architectures were chosen that could take advantage of the cheap digital

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such a system [Willenswaard 1990]. Over 1.5 megabits of memory is required. If the memory overflows, the unit switches to a PLL clock decoder. Technics will use the technology in the SH-X1000 decoder box. Technics of America has chosen not to import this unit. Technics of America refused to give me any information about the new technology or to explain why the SH-X1000 is not being imported.) The amount of memory can be reduced by *slowly* adapting the frequency of the crystal oscillator in the decoder box once the CD player starts sending data. This approach is often called a frequency-locked loop (FLL).

Designing a crystal oscillator that is both tunable and jitter-free is a difficult design problem. Cheap CD players often use master oscillators which may not be very accurate. The jitter performance of a crystal oscillator would be compromised if it were required to have enough tuning range to operate with these cheap CD players. The Wadia "RockLok" clock recovery circuit uses an FLL. The RockLok will only work with CD players having a clock that deviates a maximum of ±75 ppm from the nominal data rate. (The Technics system also requires an accurate CD player master clock, ±50 ppm, to function properly.) Wadia reports RockLok provides a 2500:1 jitter reduction over a conventional PLL-based clock recovery system.

Multiple phase-locked loops are an alternative. The first PLL is designed to capture the data free from error, without generating a jitter-free clock. The second PLL is designed to attenuate the jitter present on the clock line of the first PLL. The second PLL often uses a VCO which incorporates a tunable crystal. This approach is adopted by JVC, Kenwood, Nackamichi, and Sansui among others.

JVC strangely uses in the second PLL a VCO (74LS624) which has jitter levels comparable to the VCO in the first PLL. The JVC system does offer some jitter attenuation because the loop filter characteristics of each PLL in the system are different.

The Krell, Theta, and Wadia SPDIF decoders appear to be innovative designs. Aragon uses a similar decoder, designed for Aragon by Theta, in its D2A product. Owing to the difficulty of designing a SPDIF decoder that has low jitter, these companies were less willing to talk about the design techniques they used in the SPDIF decoder than in any other part of their design. Only Theta was willing to give figures for the peak jitter amplitude of their SPDIF decoder. Theta reports a peak jitter of less than 1 nanosecond. This 1ns peak jitter is still 2.5 times larger than the 400 ps peak jitter amplitude necessary to ensure that the full 16-bit performance of the DAC is realized [Harris 1989], [Fourre 1989]. Technics reports that their new SPDIF decoder achieves a 500 ps jitter amplitude—but, as I said, Technics of America will not be importing this state-of-the-art system into this country.

technology in order to minimize its expensive counterpart, analog technology. The most promising method involves a low-order (1 to 4 bits) DAC operating at very high interpolation rates (64 to 256, for example). A digital loop filter is placed between the digital interpolation filter and a quantizer which truncates the lower-order bits present at the quantizer's input. The output of the quantizer is returned as a feedback signal to the loop filter. A block diagram of the system is shown in Figure 11.

The oversampling architecture reduces the quantization distortion present at the output of the low-order DAC and redistributes the noise above the audible range. This process is called *noise shaping*. The quantization noise inband is reduced to levels equivalent to a 16-bit DAC. If the noise-shaping circuit has more than a single-bit output, another technique, called *pulse width modulation* (PWM), can be used to reduce the output signal to a single-level output. In a PWM DAC, the area of a pulse represents the DAC output. In a 3-bit DAC, seven one-level pulses are required to represent the 3 bits. The 3-bit PWM DAC creates single pulses at its output at 7X the word rate. The PWM DAC is inefficient and is inappropriate for a system with a large number of quantization levels. Consider, for example, a 16-bit DAC: 65,535 one-level pulses would be required. The operation of an oversampling DAC is often incorrectly described as a PWM system. The term *bitstream* describes a DAC which converts a multibit data stream to an analog signal by using a one-bit data stream. Producing a DAC that uses digital technology almost exclusively offers the following two advantages: (1) cost reduction and (2) identical performance for each properly functioning DAC. (For a given input code, the bitstream output will always be the same for each functioning DAC.) The second advantage implies that all bitstream DACs exhibit identical, good linearity performance. Properly designed, a bitstream DAC will have better linearity than a low-grade multibit DAC.

The noise shaper in the bitstream DAC is a special form of an infinite impulse response (IIR) digital filter. IIR filters can display low-level instabilities due to the truncation function of the quantizer. The low-level instabilities give rise to oscillations called limit cycles [Lipshitz 1988], [Ardalan 1987]. These oscillations will change in amplitude and frequency depending on the signal present at the input of the DAC [Dijkmans 1989]. These problems can be made worse with the use of a high-order loop filter, which is required to keep the interpolation ratio to a reasonable value [Fielder 1989].

Because the oscillations are signal-dependent, the distortion characteristics of oversampled DACs must be evaluated with multiple-tone (2–3) test signals. Traditional test signals used to evaluate audio equipment are not capable of fully characterizing an oversampled DAC. Stikvoort has found that some audible effects in noise-shaping DACs "could not be detected by just measuring them" [Stikvoort 1988]. Stikvoort found that he could not resolve audible "birdies" on his spectrum analyzer because they were "almost harmonic." He found that some pulse-train-like effects called rattle had a

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About Bitstream DAC Architecture

Currently, two groups have been working on bitstream DACs, namely NTT in Japan [Matsuya 1989] and Philips in Europe [Naus 1987]. The two DACs use different architectures. Expounding upon the relative merits of each architecture is outside the scope of this article. It should be noted, however, that the DACs are differentiated by the amount of analog technology implemented on the chip. The NTT chip implements its one-bit DAC with only two transistor switches, used in conjunction with an external passive RC or LC filter and external op amps. Philips adopts a *switched capacitor filter* (SCF) in its bitstream implementation. SCF circuit performance is dependent on: (1) the settling time, bandwidth, noise, and distortion of the internal op amps, (2) clock feedthrough and charge leakage of the FET switches, (3) linearity of the monolithic capacitors, (4) parasitic resistance and capacitance, which are unavoidable in a monolithic implementation. Moreover, fine-line CMOS processing limits power supply voltages to ±2.5 V. This is one-sixth the voltage used in bipolar analog circuits. This reduced voltage severely impacts on the performance of the DAC, especially on distortion and signal-to-noise ratios. The current state-of-the-art designs of CMOS circuits, operating at 5 V, are insufficient to allow a DAC incorporating these technologies to yield performance levels equivalent to what is achievable with bipolar ICs.

When integrated with large amounts of digital circuits, CMOS switched-capacitor circuits generate noise, coupled from the digital circuits into the analog section. This leads to IM sidebands in the region of the audio band. Furthermore, it is difficult to isolate from each other the two stereo channels integrated on the same chip, especially at 20 kHz.

Analog CMOS circuits require additional processing steps to manufacture the capacitors, thereby increasing manufacturing costs. In addition, it may not be possible, because of the special requirements of analog circuits, to use processes that allow the most compact digital circuits. This also raises costs. In telecommunications systems, the analog portions of the oversampled coders are often implemented as a separate chip, using different processes for the analog and digital chips.

Why would Philips use an SCF? The answer, I believe, is the lure of a chip that requires no external op amps. In addition, the use of a single 5 V supply (analog ground is generated internally on the chip) is attractive, since such a supply is cheaper than a ±15 V supply. Moreover, the chip could be used in portable compact disc players, which have a limited voltage supply from the batteries. The general description section of the data sheet for the SAA7320 bitstream DAC confirms this suspicion. "The SAA7320 (DAC3) is...designed for applications in low/mid-cost, portable compact disc systems." There is no question that the SAA7320 is a good device to be used for its intended applications, although its design performance (using characteristics given in the data sheet) falls far short of what can be achieved using the Philips TDA1541A-S1 and SAA7220P/B chip set.

Philips suggests the use of two SAA7320's in a differential mode in conjunction with external op amps to improve performance. This appears to be an expensive solution, since the digital filter section of the SAA7320 is then needlessly duplicated and some of the op amps on the SAA7320 are not used. Moreover, if Philips had intended the chip to be used in the differential mode at the onset of the chip's design, its internal circuitry would have been implemented with fully balanced op amps and switched capacitor circuits [Lee 1985]. It should also be noted that the design of the analog differential-to-single-ended converter can be difficult. Meridian uses a novel differential-to-single-ended converter which eliminates the common-mode input signal present in standard implementations of the circuit. Ben Duncan uses the SSM2016 in his digital decoder do-it-yourself project [Duncan 1990]. A special input stage that has true differential inputs and a very high common-mode rejection ratio is used in the SSM2016 integrated circuit. Harman/Kardon uses a discrete op amp which is optimized for a high common-mode rejection ratio. Sony uses a single inexpensive 5532 or 5534 op amp. These op amps have been shown to have poor distortion performance in the presence of a common-mode signal [Jung 1987].

repetition rate so low that they did not appear at frequencies in the audio band. Stikvoort points out that computer simulation methods which are used to design a noise-shaping DAC cannot be run long enough to show these effects. Goudie reports that the tones grow sidebands and then break up into a continuous spectrum of noise as signal level increases [Goudie 1989]. Carley states that perception of a system's sonic performance cannot be based solely on its signal-to-noise ratio if the level and spectrum of the background noise vary with input signal levels [Carley 1987].

Naus has shown that the modulator does not respond well to small-signal input changes below threshold level [Naus 1988]. Naus reports that with an input signal slightly higher than the threshold level, a gain error will introduce harmonic distortion and inband whistles. An additional problem with an oversampled DAC can occur when high-level signals are present. The signal-to-noise ratio of the DAC may decrease as signal amplitude increases because of the nonlinearity of the quantizer [Ardalan 1987]. The correct application of dither and the careful design of the loop filter are reported by all aforementioned researchers as strongly reducing the undesirable effects found in oversampling coders. Vanderkooy reports that one type of dither (rectangular pdf) is not recommended for bitstream DACs [Vanderkooy 1989]. The optimum type of dither and the placement of the dither signal in bitstream system are still the subject of research. The dither used (if any) in present monolithic bitstream chips is not likely to be optimum.

The very high quantization noise level at the output of a bitstream DAC requires a third- to fifth-order filter, in comparison with an 8× interpolated multibit system, which can use a first-order filter. The first-order system reduces the complexity of the analog section, and consequently distortion from the analog section is decreased. Modern op amps will enter slew-rate limiting if required to process a bitstream signal directly. To prevent this, a passive RC or LC network is used after the bitstream DAC and before the active circuitry. If a high-order (higher than 3) passive filter is employed, the performance requirements for the active stages can be reduced in comparison with the requirements in multibit systems. A problem with this approach is time-domain distortion from the higher-order filters. In addition, the sonic consequences of using inductors in the signal path, as would be required in a high-order passive filter, are not well documented. *[Come on, David, some of the finest loudspeaker systems use them.—Ed.]* The high out-of-band energy present at the output of the CD player may cause problems for some preamplifiers if the signals at the output of the DAC aren't filtered sufficiently. Sony, in the new CDP-X55ES and X77ES, only uses a first-order passive filter before connecting the signal to an active filter stage. The active filter stage uses an NE5532 op amp, which does not have the required settling time, small-signal bandwidth, or slew-rate specifications (see below) to operate in this application. Static THD testing has not shown any abnormalities in the Sony players, but dynamic distortion products may occur on com-

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Philips has recently introduced a modified version of the SAA7320. (This is the second modification; the first modification was the SAA7321.) The new chip, which is called the SAA7323, is claimed to offer improved idle pattern performance at low levels. The specifications for the SAA7323, in the single-ended mode, are limited and are given for typical, rather than worst-case, performance: THD at 0 dB = –90 dB; gain linearity, ±2 dB. These performance levels are between Category 1 and Category 2 performance.

Technics (MN6471M) and Sony (CXD2552) have implemented chips based on the NTT technology. Both chip sets can also be operated in the differential mode. However, unlike the Philips chip, the Japanese DACs achieve this operational mode with a single chip. In the September 1989 issue of CD Review, the Sansui AU-X911DG digital decoder/amplifier, which uses the Technics chip, was tested. *[No test of this Sansui model is being contemplated by* ***The Audio Critic****.—Ed.]* From these tests, it appears that this chip also operates at a performance level below the state-of-the-art multibit DACs. Distortion on single tones is 4 dB higher than state-of-the-art at 1 kHz and, strangely, 25 dB higher at 20 Hz. From the early reviews in the hi-fi slicks of preproduction samples of the Sony CDP-X55ES and X77ES, it appears that the Sony chip set has better performance. Sony appears to have implemented the NTT system without modification [Matsuya 1989]. The CXD2552 is manufactured in a very advanced CMOS technology. This allows the chip to run at a 45.1584 MHz clock rate. The faster clock allows a larger number of computations to occur in a given time period. This clock is not at an integer multiple of the system clock (16.9344 MHz). Sony uses an expensive frequency multiplier circuit to generate the DAC clock. (It seems that Sony spares no expense in the design of its top-of-the-line CD players—except in the analog section.) The Technics chip [Ainslie 1990] uses a modified NTT structure which has 11 quantization levels instead of 7 at the output of the quantizer. The Technics chip runs at a slightly slower 33.8688 MHz, which is an integer multiple of the system clock. Because of its higher clock rate, the Sony system interpolates at a 64× rate, while the Technics system interpolates at a 32× rate.

Both the Technics and Philips chips incorporate digital interpolating filters with a smaller number of filter taps and a smaller data path size than the digital filters analyzed in Table 3. The simplification of the digital filter is required to make room on the silicon for the circuitry associated with the bitstream DAC. The result is more passband ripple, among other problems. The Sony CXD2552 DAC does not incorporate the digital interpolation filter on the same silicon. Sony uses the excellent CXD1244 digital filter (see Table 3) in conjunction with the CXD2552. I have not seen the data sheet for the CXD2552, or for the Technics MN6471M, and thus I cannot make definitive statements about the performance of these chips. I am aware, however, that Sony has added a muting circuit to the CDP-X55ES and X77ES, which shorts the output signals of the analog stage to ground when a silent track is detected. This circuit increases the measured signal-to-noise ratio of the CD player, since the idle pattern of the DAC is not allowed to appear at the output. The circuit does not appear to perform any useful function for the consumer, and it can add distortion because the nonlinear semiconductor junction of the muting circuit is connected across the output of the CD player. Even so, the Philips SAA7323 is clearly inferior to the Sony CXD2552. Philips has meanwhile announced its very latest chip, the SAA7350, in an attempt to catch up with Sony. The SAA7350 uses a third-order noise shaper, like the NTT system, instead of the simpler second-order noise shaper. It has differential outputs, a feature the NTT system has always had, and it uses an external digital filter, as does the Sony CXD2552. Philips has not yet completed development of the 20-bit digital filter to be used with the SAA7350, but the latter can be used with the Japanese digital filters shown in Table 3. The SAA7350 continues to use switched capacitor analog circuits and a ±2.5 V analog power supply. The specifications for the SAA7350 are once again limited and given for typical, not worst-case, performance: THD at 0 dB = –93 dB; gain linearity, ±1 dB. This performance is equivalent to that of a chip between Category 3 and Category 4 and is far from state-of-the-art.

plex test signals. (For further bitstream information, see sidebar.)

Bitstream technology certainly offers superior performance in low-cost CD players that have used inexpensive, nonlinear DACs until now. I expect it to become the dominant technology in such low-cost players. The performance of the analog section in low-cost CD players is clearly compatible with the performance of the bitstream system. In mid- and high-priced CD players, I believe that a properly adjusted Burr-Brown PCM58P-K or PCM61P-K, when used with a properly designed analog section, will outperform the Philips and Technics bitstream systems, including the new Philips SAA7350. The Sony CXD2552 represents the next generation of bitstream DACs. It will be interesting to compare its performance—especially when, and if, Sony implements the analog section better than in their current products—with the next generation of multibit DACs such as the Burr-Brown PCM63P-K. Both bitstream and multibit DAC technology will obviously continue to improve in the future, and it is unclear which technology will finally prove to be superior. For the moment, only Harman/Kardon, Meridian, Sansui, and Sony are using bitstream technology in their top-of-the-line CD players. Even Philips and Technics use multibit DACs in their flagship models. Most manufacturers are using bitstream in low- and midpriced players only.

Please note that it is in the interest of the Japanese electronics industry to push the bitstream technology. The Japanese have been strong in the production of digital integrated circuits like the bitstream chips. The Japanese have been unable, on the other hand, to compete in the field of high-performance analog integrated circuits. For that reason they have had to purchase multibit DACs from the USA or from Philips. The vertically integrated Japanese companies can save a significant amount of money if they can convince the consumer that the bitstream product is superior.

Power Supplies

The power supply for the DAC and the analog stages can significantly impact on the performance of a CD player. Power supply designs for CD players vary widely. The power supplies of many of the new, low-end CD players severely compromise the performance of these players, as their power supplies have been designed to such a low price point.

Power supply rails which service the analog section must be shielded from signals originating from the digital circuitry and from the servo circuits which move the CD and position the laser mechanism [Fourre 1989]. This is the most difficult problem faced by the designer of a CD player's power supply. Three components of the power supply are critical in determining its performance. These are the *transformer*, the *storage* and *bypass capacitors*, and the *voltage regulators*. The servo circuits operate at frequencies in the audio band and require large current resources to position the laser assembly accurately. Signals from the digital sections of the CD player operate at much higher frequencies than the audio band. These digital signals make their presence felt as IM distortion components which appear in the audio band [Miller 1989]. These distortion components occur when the high-frequency signal is modulated with the audio signal in an analog stage. This gives rise to sum and difference sidebands. The difference sidebands can be in the audio range, and the sum sidebands can interact with other high-frequency components to produce secondary IM products, which may also appear in the audio band. The reduction of digital and RF noise in an outboard decoder box, in comparison with a CD player, may improve the performance considerably more than the small amount of jitter in the clock signal will degrade it.

The best method of isolating the analog and digital sections is to use a separate transformer for each section. High-end CD players such as the Yamaha CDX-1120, Sony CDP-X7ESD, and Denon DCD-3520 apply this technique. Even these players combine the ground line of the two transformers, and consequently noise on the ground line from the servo and digital sections is imposed on the analog section. Careful attention to the ground routing can greatly minimize the effect. A more sophisticated method is the use of optical couplers that isolate the analog and digital sections. This approach is championed by Onkyo, who offers dual power supplies even in the $700 Integra DX-7500. Optocouplers are also used in the CAL Tercet Mk III and the JVC K2 Interface. Both Yamaha, in its CDX-1120, and Denon, in its DCD-3520, have abandoned the optocouplers found in the previous generation of these machines. It is unclear whether this is a move to cut costs or an effort to eliminate technical problems with the optocouplers. The optocouplers can be placed either between the data recovery chip and the digital filter chip, as in the Onkyo, or between the digital filter chip and the DAC, as in the Philips LHH1000. Since the output of the digital filters is connected to the DAC inputs, and noise on these digital signal lines can corrupt the analog output of the DAC, it is desirable to have the digital signals as noise-free as possible. This is best accomplished by placing the optocouplers after the digital filter chip.

A comparable technique to separate transformers is separate windings for the analog and digital section on a single transformer. This technique is used in most midpriced CD players. An undesirable trend in many low- and midpriced CD players is an attempt to reduce transformer costs by reducing the voltage and current available at the output of the transformer. Especially disturbing is the use of transformers whose voltage output is so inadequate that only ±5 V regulated analog supply rails can be generated. The normal analog supplies are three times as large at ±15 V. Using ±5 V supplies requires the inclusion of special low-voltage analog ICs in the analog section. These analog ICs do not have the performance of ICs that run on ±15 volts because their design requires that they operate with small margins between the CD player's maximum analog output swing of ±2.8 V and the power supply rails. An example of a fairly recent CD player with ±5 V supplies is the Sony CDP-508ESD. This player also has a single transformer winding to supply both the analog and digital sections. The Sony CDP-505ESD, which was replaced by the 508, had separate windings on the transformer and ±15 V analog supply rails. The Sony CDP-508ESD has yet another cost-cutting circuit in its power supply. The 508 uses the PCM58P (unselected) DAC. This DAC requires a +12 V supply. Because the 508 has only a +5 V analog supply rail, a voltage multiplier circuit generates the DAC supply voltage. The voltage multiplier produces only an 8 V output, which is not the optimal supply voltage. In addition, the high output impedance of the voltage multiplier compromises the performance of the DAC.

Many midpriced CD players use ±12 V supplies (e.g., the Sony CDP-608ESD and Kenwood DP-8010). While the impact on sound quality is not as severe as with ±5 V rails, it appears to me that it is more sensible to retain the better power supply and eliminate trivial features, such as nonvolatile memory for the programming functions or remote volume controls. Not all manufacturers have compromised their transformers. For example, the $429.95 Philips CD-60 has full ±15 V analog supply rails.

The *active voltage regulator*, which is placed after the unregulated power supply rails, creates a supply rail voltage set to a precise value. The regulator suppresses ripple and noise found on the unregulated supply. The regulator output has a low output impedance which maintains a constant voltage on the regulated line under changing load conditions and prevents noise from coupling into the regulator's supply line. Separate regulators on the supply lines for the DAC and analog sections isolate these circuits from each other. Ideally, each power supply terminal on the DAC (2 for the PCM58P, 3 for the TDA1541, and 4 for the PCM56P and PCM61P) should have a separate regulator. Since the signal at the output of the digital filter is connected to the DAC, a separate regulator for the digital supply to the digital filter chip (separate from the regulator which supplies the other digital circuitry in the CD player) is advantageous so as to minimize signal interference. An additional regulator may be used in the crystal clock oscillator circuit to prevent supply noise from modulating the oscillator (this would result in clock jitter). Improvements in stereo separation may possibly be achieved with a dual mono configuration that includes separate regulators for the left and right channel analog sections. It is also possible to use separate regulators for the left- and right-channel DACs, in the case of DACs which process only one channel of the stereo pair, such as the Burr-Brown PCM56P, PCM58P, and PCM61P, and the Analog Devices AD1856 and 1860. Dual mono configuration cannot be achieved in the TDA1541 since a single chip houses both the left- and right-channel DACs. Players are often advertised as dual mono or twin mono, even though both channels share the same power-supply regulators. The 1989 brochure for the Sony ES line, for example, claims that the CD players are twin mono, although both channels use the same analog supplies. This mistake is possibly the result of changes incorporated after the preparation of the brochure. Only an examination of the service manual will indicate whether or not the player has the required regulators for dual mono operation. The total number of regulators can be high, usually in the low to middle teens. The CAL Tercet Mk III uses 23 regulators. On the other hand, the number of regulators used in some Japanese CD players is being reduced. For example, the discontinued Sony CDP-505ESD had 7 regulators (including 2 for the servo system), while the CDP-508ESD that replaced it uses only 3 regulators.

Like op amps, regulators can either be purchased as integrated circuits or implemented with discrete resistors, diodes, and transistors. The 78XX, 79XX, 317, and 339 are the most popular devices. The 317 and 339 are frequently cited by American high-end manufacturers as offering better performance than the 78XX and 79XX devices. Rarely are these regulators seen in Japanese equipment because they require external components. Integrated regulators of a given device type are available in a variety of different current ratings. Regulators with high current ratings have a lower output impedance, but they are more expensive. A well-designed discrete regulator will have lower output impedance, especially at higher frequencies, when compared with integrated devices. Moreover, when compared with integrated devices, discrete regulators can have lower output noise, higher ripple rejection, better temperature stability, and better regulation under load [Marsh 1983], [Breakall 1983]. A high-performance regulator is often formed in two stages, with a preregulator near the power supply and a set of subregulators, connected to the preregulator, near the active analog circuits [Didden 1987]. Integrated regulators can sometimes be incorporated in the preregulator or the slave section (but not both) in high-performance regulating systems without significant performance degradation. The vast majority of American CD designers and modifiers use all discrete regulators or a combination of discrete and integrated regulators in their CD players.

Some Japanese and European manufacturers also adopt discrete primary regulators in their high-end CD models (e.g., Philips CD-80, Sony CDP-608ESD, and Sony CDP-X7ESD). The regulator stage in the Pioneer PD-71 uses a very interesting regulator circuit. The regulator has a complementary class AB output stage. Normally, regulators are designed to either source or sink current and consequently only an NPN or PNP transistor is found at the regulator's output. A standard positive regulator circuit, for example, presents a low impedance output only when it sources current. In the presence of large-amplitude, high-frequency symmetric noise sources, a complementary regulator may reduce the amount of noise which couples to the regulator's output line.

The size of the storage capacitors used after the power transformer determines the ripple voltage and current reserve of the power supply. This can vary from hundreds of microfarads to tens of thousands of microfarads. The shunt capacitors placed on the regulated power supply rails ensure that a low source impedance is present at frequencies at which the active regulator stage's impedance is increasing. Many manufacturers combine low-ESR electrolytic capacitors, film capacitors, and ceramic capacitors in parallel to achieve low shunt impedances to noise present within the wide spectrum of frequencies inherent in a CD player. These high-quality capacitors, which must be used with each active regulator, can cost several dollars each, even in high quantities. Hence, their use can significantly affect the final price of a CD player. Unfortunately, some high-end audio manufacturers claim mystical properties for their private-label capacitors. There is little correlation between the brand of capacitor used in a power supply and the power supply's performance. The size and type of capacitor makes the performance difference. Capacitors are not a panacea. Various small modifiers of audio equipment, who do not have solid engineering foundations, will add expensive capacitors across almost any power supply point to be found on a CD player. Since these modifiers are unable to perform careful engineering analyses to demonstrate what types and values of capacitors are best suited for each circuit, the changes that result are often too superficial to affect the sound of the player. Mark Brasfield of MSB Technology (a degreed engineer working at Stanford Research Institute) takes an interesting minority approach which seeks to eliminate as many bypass capacitors as possible. Mr. Brasfield designs the voltage regulators to present a very low impedance to the active circuitry, even at very high frequencies.

For cost control reasons, some of these midpriced players show smaller storage capacitors than earlier designs. The Sony CDP-950, for instance, has only a 1000 µF capacitor on the unregulated negative analog supply node, while its predecessor, the CDP-910, used a 3300 µF capacitor. Transformers, regulators, and filter capacitors are good sites for compromise from a marketing point of view, as these changes are not visible to the consumer in the showroom.

Analog Stages in CD Players

Let us say you own a Sony CDP-X7ESD, for which you paid $2000 (minus whatever discount was available). That bought you a player with one of the most advanced transports in the business. Your player also has separate transformers for the analog and digital sections, and the power supply has discrete preregulators for the analog section and nine integrated subregulators. In addition, the CDP-X7ESD has a special-selection grade of the Burr-Brown PCM58P, exclusive to this unit. Unfortunately, you also have three op amps in the signal path, with a price of much less than $1 per chip in the quantities purchased by Sony. Do not feel too distraught; at least you did not purchase a Sony CDP-R1/DAS-R1 for $8000 or a Philips LH1000 for $4000, both of which use $1 op amps in their respective analog stages.

It is one of the great mysteries of high-end hi-fi why the large manufacturers shun modern discrete amplifier design techniques in their high-end products. The audio performance of a CD player could be greatly improved if the inexpensive op amps were replaced with discrete amplifiers. The problems with inexpensive op amps and the proper design of discrete amplifiers is beyond the scope of this article. A well-designed discrete amplifier offers many advantages [Marsh 1985], [Borbely 1987], [Howe 1989], [Jensen 1980], seven of which will be cited here.

First, the gain and transfer response of the individual stages and the complete amplifier can all be adjusted, as can the operating current of each stage. This allows the optimal trade-off of noise, open-loop distortion, bandwidth, and large-signal settling time for a given application [Cherry 1982]. Second, the return loop gain, which sets the amount of feedback of the amplifying stage, is adjustable. Refer to [Otala 1980] for a rigorous analysis of the detrimental effects of excessive feedback. The validity of the analysis when applied to analog amplifiers has been questioned, however [Cordell 1983], [Cherry 1983]. Third, designing an output stage with class A operation, no I/V current limiting, and low output impedance becomes an option. Fourth, the designer can select from the thousands of discrete transistors available and use transistors with different operating parameters in different stages of the amplifier to optimize performance. In addition, bipolar, JFET, and MOSFET gain elements (and tubes for those who must have a device that glows in the dark) may be mixed in one amplifier topology. Fifth, passive components of any type or value are practical. This is of particular importance when designing an amplifier with low open-loop distortion, good power supply rejection ratio, and good transient performance. Sixth, there is the option to use power supply rails of more than ±15 V. Seventh, there is the option to use fully complementary topology to reduce open-loop harmonic distortion. If the open-loop distortion is lowered, the amount of negative feedback can be reduced by the same amount without changing the closed-loop distortion. A disadvantage of discrete circuitry, when compared to an IC or hybrid, is increased parasitic capacitance from the

Designing the Analog Stage

Analog stage design in a CD player is similar to preamplifier stage design. The functional first section of the analog stage, namely the current-to-voltage converter, is the most difficult to design. The current-to-voltage converter takes the current output of the DAC as its input and generates a voltage signal at its output with a level linearly proportional to the current entering the stage. Design problems arise in this stage because the DAC output is not a bandlimited analog signal, but rather a set of current steps changing at the word rate of the DAC. The stage must respond to the step change in current with a step change in voltage. The voltage change must occur within a small percentage of the conversion period, and the voltage output must settle to the new signal within an accuracy of 0.0015% (for 16-bit accuracy) of the output expected from an ideal current-to-voltage converter. To achieve this level of performance, the amplifier must have a high slew rate, a wide bandwidth, linear operation with "large-signal" inputs, and large stability margins. As an example of the requirements in a current-to-voltage converter, consider the required slew rate for an amplifier to slew completely from the maximum positive signal to the maximum negative signal (a swing of 5.6 V typically) in one-tenth the conversion period of the DAC. For an 8x interpolating system this is 285 ns. A slew rate of at least 20 V/μs is required.

To settle within 0.0015% of a signal's final value, a circuit with a single-order lowpass response must settle in 11 time constants. This requires the current-to-voltage converter to have a bandwidth 11 times the word rate. For an 8x interpolating system, this requires the current-to-voltage converter to have a bandwidth greater than 4 MHz. To ease these requirements somewhat, a first-order filter function, which forms part of the reconstruction filter, can also be incorporated in the current-to-voltage converter. An additional operating constraint requires that the input of the current-to-voltage converter be held at its ground potential so that the DAC is terminated into a virtual short. If the output of the DAC is not held at a constant ground potential, the linearity of the converter is degraded. The current-to-voltage converter should also show good rejection of high-frequency noise, which may be present on the power supply rails. A current-to-voltage converter stage should have a minimum power supply rejection ratio of 60 dB at 100 kHz.

A current-to-voltage converter can be crafted from a voltage amplifier [Jung 1986], though the performance criteria are more easily met using a current feedback (transimpedance) amplifier [Evans 1985], [Goodenough 1987]. This topology offers higher slew rates, greater bandwidth, and lower settling times than is possible with an op amp topology. This circuit is more easily understood than the op amp circuit. It, therefore, will be used as a point of reference in explaining the operation of a current-to-voltage converter stage. The transimpedance amplifier (Figure 12) works by placing a low impedance across the input to act as a short. This forces the input voltage V_{in} to be held at ground potential as required for proper D/A performance. Feedback, in the form of current, is applied to the input through R. Current from the DAC and current from the feedback branch enter the short-circuit input of the transimpedance amplifier.

The gain stage, whose transfer function is A (with units in ohms), senses the current in the short and transforms it into a voltage. If A is large enough, the feedback loop will be satisfied when the current feedback is equal to the input current. Since the input node is held at ground potential, the voltage output of the stage is the voltage across the resistor. Hence,

$$V_{out} = -I_{in}R$$

This is the ideal equation for a voltage-to-current converter with conversion impedance R.

If the designer has chosen to reduce the amount of negative feedback in an audio amplifier, the transimpedance amplifier has an additional advantage. Cordell argues that an amplifier can be designed with high global feedback rates and avoid exhibiting dynamic distortion [Cordell 1980]. D. C. Wadsworth of Phototronics contends that this argument is valid only if the input signal is bandlimited to the audio band. The input to a current-to-voltage converter has a bandwidth well into the megahertz region, thereby violating this condition. A voltage amplifier, when used in a current-to-voltage converter, requires high rates of feedback to reduce the input impedance of the amplifier [Millman 1979] to a point at which the DAC output will not move significantly from ground potential. Only transimpedance amplifiers and current amplifiers—see [Didden 1989] for an elegant treatment of a current amplifier as part of an I/V converter stage—are feasible if a low rate of negative feedback is to be used around an amplifier which simultaneously provides a low input impedance. The principal disadvantage of current-mode amplifiers is that they are not as easy to use as op amps [Goodenough 1990]. Increased noise levels are also a potential problem with current-mode amplifiers.

Accuphase, Precision Audio, Barclay Audio (a company no longer producing complete CD players), and M. S. Brasfield adopt transimpedance amplifiers in their respective current-to-voltage conversion stages. Interestingly, the latter three are small large PC board traces. This reduces the speed and bandwidth of discrete circuits and increases settling time. Companies using discrete amplification include CAL (Tercet Mk III), Aragon, Harman/Kardon, and Krell.

Latitude in discrete amplifier design means that a designer can create an amplifier that will impose almost no audible distortion on the signal or, alternatively, an amplifier that is distinctly inferior to a $1 op amp. Discrete amplifiers are expensive to build properly. Hence, op amps play an important role in low- and midpriced CD players. An analog back end cannot solve problems in the stages that precede it. I believe that a CD player that incorporates an op amp analog stage and the highest-quality DAC available will outperform a CD player with a less expensive DAC and a discrete analog stage.

It is possible to use op amps and other monolithic functional blocks, such as buffer circuits [Williams 1986], in conjunction with a few discrete transistors to form an amplifying stage that will approach the performance of a discrete amplifier. Companies adopting this approach are AVA, Proceed, Pioneer (PD-71), and PS Audio.

Performance requirements for an amplifier in the sample-and-hold stage which follows the current-to-voltage converter are even more difficult to meet than for the current-to-voltage converter itself, if the DAC must be deglitched. The settling time is the culprit here. The amplifier in this stage must settle in half the time of the current-to-voltage converter (the circuit is holding the data for half a clock cycle). In addition, it must have a high input impedance and show stability with capacitive loads. Often, more than one amplifier is needed to meet all the requirements. Additional considerations in the sample-and-hold circuit are the accuracy and speed of the transistor switches. Furthermore, the performance of the storage capacitor, where the data is held, is critical [Jung and Marsh 1980]. None of the sample-and-hold circuits used in CD players that I have examined are sophisticated enough to meet these performance requirements. The most practical solution in the design of a high-performance CD player is the elimination of this stage by using a DAC with low glitch current.

The last stage in the analog section

American companies with degreed electrical engineers heading the design department. All three companies developed the solution independently. Small CD modifiers who do not have an engineering background continue to use op amps in their designs. Unfortunately, no matter how many different types of op amps they try, they will never overcome the fundamental design problems understood by trained professionals.

A designer who has chosen to use integrated circuits in the analog stages now has the option of using an integrated circuit designed specifically for high-end and professional applications. The chip is the Photo-tronics PA630 current conveyor, designed by D. C. Wadsworth [Wadsworth 1989]. The current conveyor (patents pending) is a special form of current amplifier that requires no global feedback. Output buffers included on the chip for filtering functions (see below) also use no global feedback. The chip is processed using an advanced (and expensive) complementary bipolar process. All passive components are left off the IC so that high-quality discrete passives can be used. No short-circuit protection, which could introduce nonlinearity and increase settling time, is included. Because the entire circuit is implemented on a single chip, the rise time of the current conveyor is less than 25 ns. Since no global feedback is used, THD is an order of magnitude (0.02% at 0.5 V rms) higher than that for circuits that use even moderate levels of global feedback. The level is still low enough to be almost certainly inaudible. The chip also has fairly low power supply rejection; consequently, it requires good supply regulation. These chips are priced higher than standard op amps because of the advanced processing technology. In addition, new monolithic devices must be priced high enough to cover the cost of their development phases. Older IC devices are priced closer to the direct cost of manufacturing. The price of the PA630 will restrict its use in budget machines, but midpriced machines should be able to incorporate these devices.

A number of high-performance monolithic current feedback amplifiers have been recently introduced to the market. The principal specifications for these op amps are given in Table 4. Unlike the PA630, these chips are not specific to CD players. They offer lower THD levels, though dynamic distortion products may result. Two provisos: these wide-bandwidth IC devices can be difficult to work with and will oscillate if not properly employed. For this reason, I do not suggest that you attempt to replace the op amps in your present CD player with current feedback amplifiers.

A few low-noise, high-slew-rate, wide-bandwidth monolithic operational amplifiers can be used with good results in the current-to-voltage conversion stage if a designer chooses to implement the current-to-voltage converter with op amps. Jung uses the op amp's offset adjustment pins in an innovative manner to linearize the first stage of some op amps [Jung 1986]. This raises the linear input range and lowers the global feedback rate by forming an inner feedback loop around the first stage of the amplifier. A relatively inexpensive Signetics NE530 op amp is used so that the circuit can create a current-to-voltage converter with good performance in a popularly priced CD player [Jung and Childress 1988]. A popular op amp used in some high-end CD players as a current-to-voltage converter is the PMI OP42. The performance of the OP42 and of some more recently introduced op amps is given Table 4. As can be seen, the OP42 is an excellent choice for cost-sensitive applications. Table 4 also gives the performance data for the NE5532, NE5534, and LM833 op amps used in most European and Japanese CD players. As can be seen from the chart, these devices do not meet the 20 V/µs slew rate requirement calculated above. Furthermore, the critical specification of settling time is not disclosed for these chips. Manufacturers often state that they are forced to use these op amps because they have lower noise than higher-speed devices. As can be seen from the chart, this objection is substantiated, with the exception of the Burr-Brown OPA627. The reduction in the signal-to-noise ratio that would result if the other high-speed op amps were employed is an insignificant 9 to 12 dB—insignificant because all of them exceed the inherent signal-to-noise ratio of a 16-bit digital encoding system, which is 98.1 dB and therefore the effective limiting parameter. If this reduction is not acceptable, a high-speed op amp can be used in conjunction with a simple discrete preamplifier to form a low-noise, high-speed amplifier. As an example, the data sheet for the SSM-2210 transistors shows an amplifier with a 1.7 nV/$\sqrt{\text{Hz}}$ noise level, 40 V/µs slew rate, and 63 MHz bandwidth. The noise level of this amplifier is lower than that of any commercial op amp.

Mike Moffat of Theta uses the PMI OP42 operational amplifier in his designs. Mr. Moffat claims that the use of Teflon PC boards (which are extraordinarily expensive) makes a significant difference in sound quality. *[Ahem.—Ed.]* A discrete design would take up too much board space to allow the use of a Teflon PC board, Mr. Moffat argues. Mr. Moffat states that an op-amp-based analog stage will outperform a discrete stage on a cheaper PC board. Mr. Moffat also states that a current feedback amplifier, while exhibiting fast settling to 10- or 12-bit accuracy, may not settle as quickly to 16-bit accuracy as the OP42.

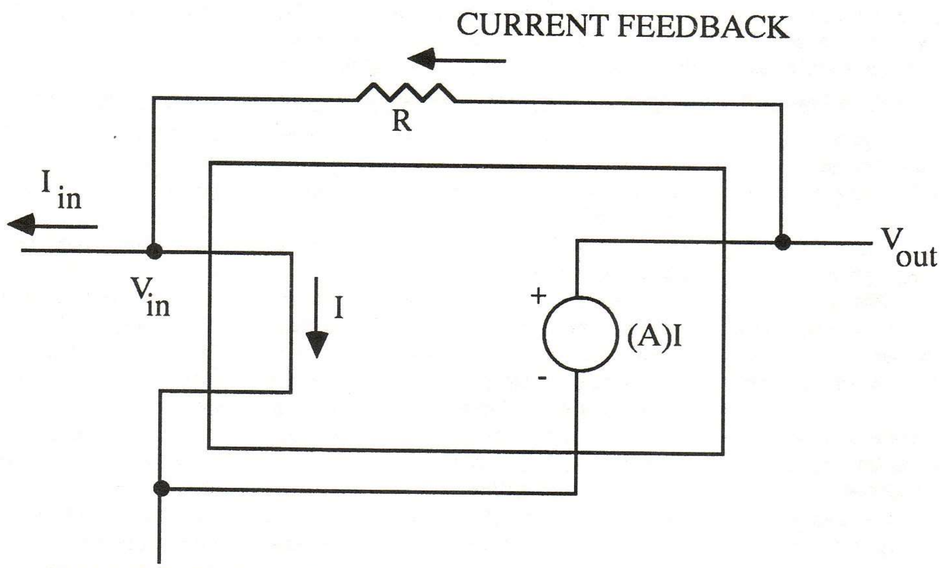


Figure 12

of a CD player is the reconstruction filter stage. This stage removes the high-frequency spectral components inherent in a sampled data system. The use of digital interpolation reduces the required order of the analog filter stage considerably. For an interpolation rate of 8×, a simple one-pole filter is adequate, although most Japanese players (Sony, Denon, Pioneer) use a third-order filter. The Onkyo DX-7500 offers the option of a first-order or a third-order filter. The Yamaha CDX-1120 has this option; however, a bipolar analog switch switches the filter in and out of the circuit instead of separate jacks for the two options. Only the passive components are bypassed in the CDX-1120, not the additional active stage. It is possible that the bipolar analog switch adds more distortion than is removed by shorting out the passive components. The higher-order filter is required if the preamplifier stage following the CD player is not linear outside the audio band. Most modern preamps should have no problem with the low-order filter. Several designers of CD playback equipment disagree, finding that some otherwise excellent preamplifiers and power amplifiers require greater attenuation than a first-order system can provide. For this reason, Aragon uses a third-order Bessel filter. As stated above, the first-order filter can be incorporated in the current-to-voltage converter. Consequently, a single amplifying stage is used for the entire analog signal-processing section. In addition to the Onkyo DX-7500, the Krell, PS Audio, and Theta digital decoders have a single analog stage. The Wadia goes even further, incorporating only RF filtering.

If the output stage of the I/V converter is not robust enough, the settling performance of the I/V converter can be affected by the loading of the preamp and cables. This problem is most likely to occur with an I/V converter that uses only an integrated circuit. To prevent the settling time degradation, Theta and Wadia incorporate an additional buffer stage after the I/V converter.

It is not possible to combine the current-to-voltage converter and the filter in one stage for all DACs. The problem arises in DACs with mismatched full-scale current outputs (Burr-Brown PCM56P, PCM61P, Analog Devices AD1856 and AD1860). The mismatches are caused by processing variations in the values of the resistors at the DAC core. To match the voltage output between the two stereo channels at the output of the current-to-voltage converter, an additional monolithic resistor—which tracks the value of the resistors in the DAC's core—is used, since it is on the same die. When this resistor is incorporated in the feedback loop of the current-to-voltage converter, the effects of the processing variations are canceled. Unfortunately, the absolute value of the feedback resistor varies between different dies, so it is incapable of forming a filter with a precise time constant. The current-to-voltage converter and first filter stage cannot be combined because of this, and an additional active gain stage is needed. CAL, Madrigal, Theta, and Kinergetics adopt the Burr-Brown PCM61P by carefully matching the current output of pairs of the DACs to obviate the need for an internal feedback resistor. This approach requires significant testing, and some working devices may not be usable if they cannot be matched to another DAC. All four DACs listed above have a built-in op amp which can form the current-to-voltage converter. The op amp on the PCM56P was not adopted by the Japanese in their mid-priced and high-end products. The op amp on the Analog Devices may have better performance, since it is implemented in a more advanced (BICMOS) technology. The Burr-Brown PCM58P has a laser-trimmed current source so its current output is matched between devices.

The Philips TDA1541 has matched current outputs because both channels are on a single die. Unfortunately, the device can only work at a 4× interpolation rate, thereby requiring a third-order filter. To implement a third-order filter, two active stages, including the current-to-voltage converter, are required. Mark Brasfield only uses a first-order filter in his TDA1541-based design. This creates a significant amount of high-frequency output from his player, centered around 176.4 kHz. Mr. Brasfield states that the level of these out-of-band signals is sufficiently low to render them inconsequential.

Sony, Denon, and Onkyo use GIC-based third-order filters in their more expensive players. The GIC (generalized impedance converter) filter creates high-order (fifth or greater) filters which have low sensitivity to passive component variation. Accuphase used the GIC filter in its first CD player (the player did not use digital interpolation) to implement a high-order filter and achieved good results. Subsequently, the GIC circuit appeared in other CD players with low-order filters under the assumption that the performance advantage of the GIC circuit would be retained. It is argued that the GIC circuit is not in the path of the analog signal as in the more standard Sallen-Key topology. The argument is flawed because a unity-gain buffer must follow the GIC filter. The Sallen-Key topology uses an identical unity-gain circuit, and the reactive components around the Sallen-Key filter are not in the circuit at audio frequencies. The GIC implementation requires two additional op amp stages and four additional passive components.

A preferable approach would save the cost of these components by building a Sallen-Key stage with an improved op amp. Pioneer, in their PD-71, and Kenwood in the DP-8010 use a filter section which is formed around an op amp in the inverting configuration. This eliminates the common-mode distortion of an op amp when it is used in a buffer configuration. The disadvantage of this circuit is the added inverting stage which is required if absolute polarity at the output is to be retained. This can be accomplished with a simple digital circuit, though Pioneer chose an analog circuit. The circuit used by Pioneer and Kenwood also has the additional disadvantage of a low input impedance, which may be difficult for the previous stage to drive.

The electrical requirements of the active voltage gain stage in a filter stage are similar to those of a preamplifier stage, except that the bandwidth of the stage should be wide enough to ensure proper operation of the filter in its stop-band region. The PSRR (power supply rejection ratio) of the amplifier should be high at frequencies outside the passband, so that noise on the power supplies is not coupled to the output.

The Sallen-Key filter circuit does not require a stage with voltage gain. Hence, a simpler unity-gain buffer can be used. Economics favor the design of a discrete buffer when compared with a discrete voltage gain stage. If it is uneconomical to use a discrete circuit in the filter section, a high-quality monolithic op amp or buffer should be used. Jung and Childress discuss the performance of a variety of op amps and recommend those that yield the best performance [Jung and Childress 1988], [Jung 1987]; they also also discuss monolithic buffers [Jung and Childress 1988]. In addition to the active stage of the filter, the passive components of the filter can also affect sound quality [Jung and Marsh 1980].

The output from the reconstruction filter has a DC offset originating from both the active gain stage and any random or systematic DC offset in the DAC and/or the digital filter. The problem is not unique to CD players; preamplifiers and amplifiers also will have DC offsets at their outputs. This offset can be eliminated with a coupling capacitor. However, great care must be taken to ensure that the capacitor does not affect the sound quality of the signal. High-quality capacitors can be expensive and may not be available in large enough values to ensure a low enough cutoff frequency to provide the best possible bass response. As with the power supply, bypass capacitors and multiple capacitors of different types and values may be combined to yield the most transparent sound possible through the DC blocking stage.

The alternative to a DC blocking capacitor—used by Theta, Precision Audio, Philips in the CD-80, and Onkyo in the DX-7500, among others—is a DC servo that nulls out the DC component from the output by placing a compensating DC voltage at the input of a stage that precedes the output [Clark 1982]. Unlike a coupling capacitor, a DC servo is, itself, not in the signal path. Hence, it does not affect the sound quality, provided it is designed properly. The DC servo often is incorrectly described as allowing frequency response down to DC. The servo actually does not allow DC or very low-frequency signals to pass to the output. If the DC servo fails, or a power supply rail collapses, or a circuit in the forward path of the servo fails, a 15 V or higher DC voltage will appear at the output of the CD player. A power amplifier or loudspeaker can be readily destroyed if this failure mode occurs. A fail-safe protection circuit can be included, though only Philips among the above manufacturers has included the protection circuit. Jon Schleisner of Precision Audio argues that the chances of failure of the circuit are small. Further, he suggests that a large coupling capacitor provides a time constant of sufficient length to allow a 15 V pulse of over a second's duration to appear at the output of the CD player if the power supply or active circuitry should fail. This pulse could be of sufficient amplitude and duration to damage a power amplifier or speaker.

The connection of the output of the DC servo to the analog stages in a CD player presents some difficulty when compared to its use in a preamplifier or power amplifier. Jung argues that the DC servo should not be terminated at the junction between the output of the DAC and the input of the I/V converter if the TDA1541 DAC is used [Jung and Childress 1988]. Jung reasons that the DC output drift during DAC warm-up and low-frequency noise present at the output of the DAC will cause problems. In contrast, Precision Audio and Philips, among others, terminate their servo at this junction and have reported no problems with the DC servo circuit.

A CD may be encoded with digital data either flat in frequency response or emphasized with a high-frequency boost. The latter encoding requires a de-emphasis filter at the filter stage so that the disc is played back with a flat frequency response. This additional filtering reduces the quantization noise present at the output of the DAC. The noise arises when the original data is quantized to 16 bits in the recording process. The de-emphasis filter is usually implemented by placing a passive network across the feedback resistor of the current-to-voltage converter or the first filter stage. This approach is problematic, since a solid-state device is used as a switch and, consequently, nonlinear junction impedances of the device can affect the sound quality when the switch is off. The problem can be remedied by using a relay. Regrettably, in this application the relay must be of a high-quality design if reliability problems with the relay contacts are to be prevented [Duncan 1988]. Some of the relays I have seen in CD players are not of high enough quality to overcome these reliability problems. Another problem with the placement of the de-emphasis network across the feedback resistor is instability arising from the reactive components in the network. This can be resolved only by reducing the open-loop bandwidth of the amplification stage. A passive de-emphasis network placed between the current-to-voltage converter and the first filter stage has been employed by some designers to avoid compromising the amplification stage. A single analog stage cannot accommodate a passive de-emphasis circuit. Mark Brasfield believes the disadvantage of placing reactive components in a feedback loop outweighs the advantages of removing the second amplifier stage. Brasfield's CD players perform all filtering, including de-emphasis, passively so that no reactive components are in the feedback loop. The Phototronics PA630 chip also uses passive filtering and de-emphasis.

The NPC SM5803 and Sony CXD1244 digital filters include an optional IIR filter to perform the de-emphasis function. This eliminates the analog components required for the function. The disadvantage of performing the filtering in the digital domain is that quantization noise is still present at the output of the filter, since the signal has not yet been converted to the analog domain. This means that the advantage of the de-emphasis circuit is not fully realized. The problem becomes less significant for DACs with 18-bit linearity.

The final design consideration is the muting circuit at the output of the CD player. This circuit prevents large voltage pulses from appearing at the output of CD player as it is powered on and off. These pulses can destroy an amplifier or speaker if they are not suppressed. The muting circuit places a short across the output of the CD player. This can be accomplished with a transistor or a relay. The relay is preferred, since the transistor will present a nonlinear impedance to the output when the muting circuit is not activated. The relay and its associated circuitry (which drives the relay) are obviously more expensive than a transistor switch. The relay, therefore, is found mostly in high-priced CD players. One exception is the midpriced Onkyo DX-7500. Some modifiers eliminate the transistor switch but do not replace it with a relay. They short out the power switch, letting the unit run continuously. This is a risky move, since a power interruption will cause pulses to appear at the CD player's output. If you are using a passive preamp and a power amp with a large supply filter bank, these pulses will appear at your speaker terminals and possibly destroy your speakers.

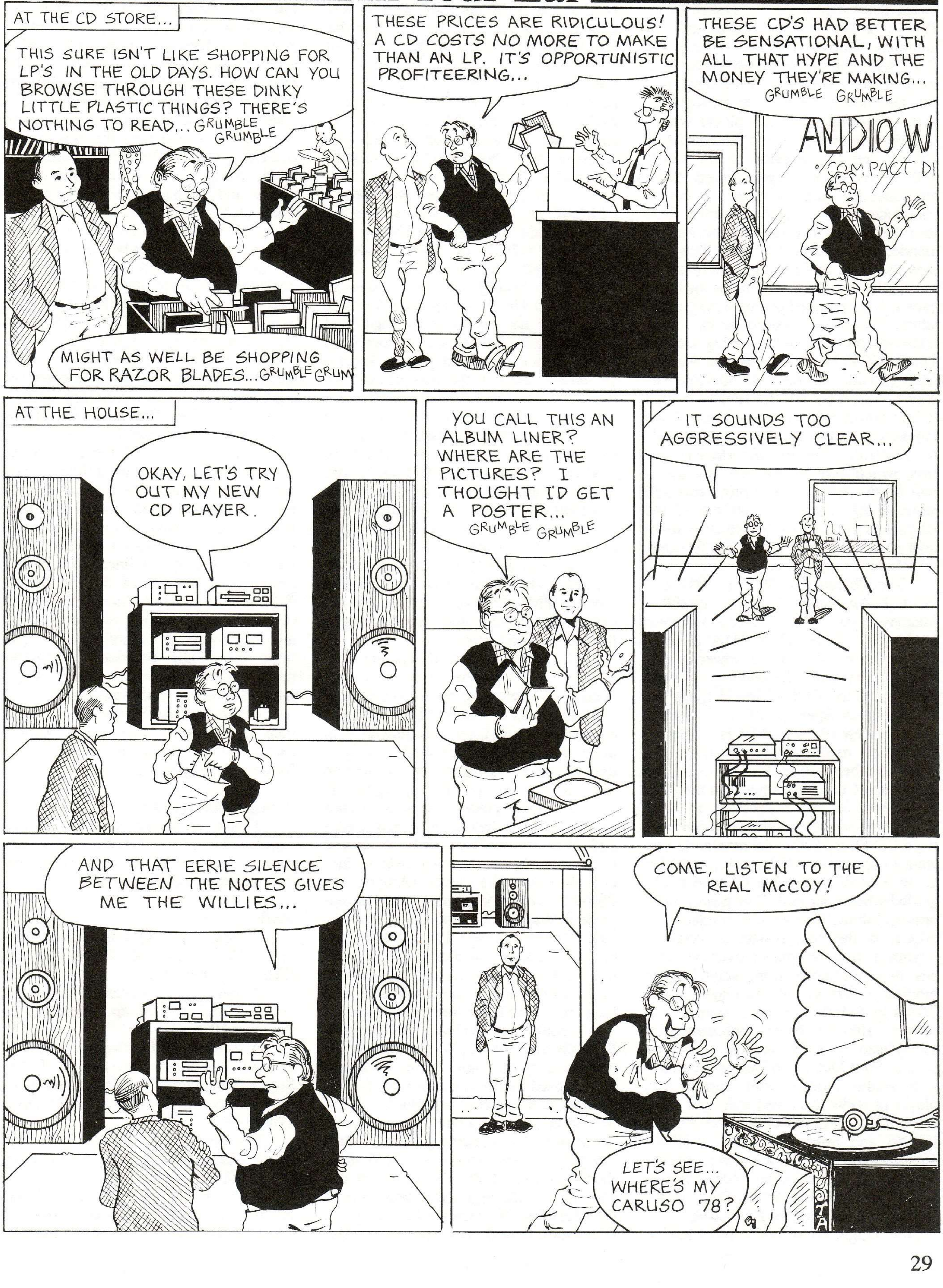
For the best possible performance, it is desirable not to load the output of the CD player with a remote volume control circuit. Eliminating the remote volume control—as well as a headphone amplifier—helps keep PC board traces as short as possible, improving stereo separation. Eliminating rarely used features also lowers the cost of the player. Many modifiers disconnect these functions during the modification process, and high-end CD players generally do not include them. Surprisingly, the Pioneer PD-71 also does not have these features—an unusual example of engineering considerations winning out over marketing considerations. Finally, four words regarding the remote volume control function: do not use it! These circuits are built to very low price points with cheap potentiometers and IC-based amplifiers. The line stage in your preamp should offer much better performance. If this is not the case, the purchase of a new preamp might prove worthwhile.

Recommendations

The October annual equipment issue of *Audio* magazine lists 80-odd different manufacturers and modifiers of CD playback equipment. Obviously, I have examined only a small sampling of these machines. This is particularly true in the low-priced category.

One CD player I can recommend is the **Magnavox CDB-630.** The CDB-630 has a list price of $399.95 but is heavily discounted. This machine offers a mid-grade (A) selection of the Philips TDA1541, a ±15 V power supply, and a well-designed CDM-4 disc transport. While the CDB-630 is a good value at its price point, the plastic chassis, single-sided phenolic printed circuit boards, extensive use of surface-mount components, and an abundance of plastic components in the CDM-4 are indications

In Your Ear


AT THE CD STORE...
THIS SURE ISN'T LIKE SHOPPING FOR LP'S IN THE OLD DAYS. HOW CAN YOU BROWSE THROUGH THESE DINKY LITTLE PLASTIC THINGS? THERE'S NOTHING TO READ... GRUMBLE GRUMBLE
MIGHT AS WELL BE SHOPPING FOR RAZOR BLADES... GRUMBLE GRUM
THESE PRICES ARE RIDICULOUS! A CD COSTS NO MORE TO MAKE THAN AN LP. IT'S OPPORTUNISTIC PROFITEERING...
THESE CD'S HAD BETTER BE SENSATIONAL, WITH ALL THAT HYPE AND THE MONEY THEY'RE MAKING... GRUMBLE GRUMBLE
AT THE HOUSE...
OKAY, LET'S TRY OUT MY NEW CD PLAYER.
YOU CALL THIS AN ALBUM LINER? WHERE ARE THE PICTURES? I THOUGHT I'D GET A POSTER... GRUMBLE GRUMBLE
IT SOUNDS TOO AGGRESSIVELY CLEAR...
AND THAT EERIE SILENCE BETWEEN THE NOTES GIVES ME THE WILLIES...
COME, LISTEN TO THE REAL McCOY!
LET'S SEE... WHERE'S MY CARUSO 78?

that the reliability of this player will not match that of the more expensive units. The CDB-630 is a favorite of the CD modifiers. Some modifiers replace the DAC with the S1 selection grade (and some do not!), add new power transformers, and nearly rebuild the analog section. The result is often a player with a four-digit price tag. Given the quality of construction and mission of the CDB-630, I recommend you avoid these modifications and purchase a better-constructed CD player. Another low-priced CD player which can be recommended is the **Rotel RCD855.** This unit uses the same basic parts as the Magnavox CDB-630, but according to the company it has upgraded power supply regulators, passive components, and improved op amps in the analog section. The RCD855 has fewer features than the CDB-630 and lists for $349.00. Rotel declined to send the unit for review. A final unit to consider in this price range is the **Harman/Kardon HD7500,** which sells for $449.00. *[However, the Mark II version, which may be the only one available by the time you read this, costs $80 more.—Ed.]* A complete review of the Harman/Kardon HD7600, which is identical to the HD7500 except for features, is included in this issue.

The **Philips CD-80** ($799.95) is a much better value than any modified Magnavox can offer. The best selection grade (S1) of the TDA1541A is combined with an all-discrete power supply regulation system. The unit uses a metal chassis and the CDM-1 Mk II transport, which has far fewer plastic components. A full review of the CD-80 is in this issue. Most modifiers give nebulous excuses for their preferences in using the Magnavox CDB-630 rather than the Philips CD-80 as the foundation of their modifications. You might consider buying a CD-80 if $800 is the maximum you want to spend on a CD player, and later go to a modifier service to have it upgraded when your cash flow position improves. I think this two-step purchase approach is the best reason to consider buying a modified player over one that has been designed from scratch. The modifier offers you a CD player in the $1000 to $1500 price class even if you cannot afford it in one purchase. You might also consider having a modifier such as Paul McGowan Designs or Precision Audio modify your current CD player provided it is well-built and has a linear DAC. The **discontinued Philips CD960** and **Sony CDP-910, 705ESD, 605ESD,** and **505ESD** are also good candidates. If you have an older Philips-based player with a 14-bit DAC (CAL Tempest I, Kinergetics KCD-20A, etc.), Digital Upgrades offers a modification which replaces the digital filter with the NPC SM5813 and the DAC with the Analog Devices AD1860-K. At $300 this upgrade is a good value for owners of these older CD players, provided the transport mechanism is in good condition. Given the quality of performance available from the PS Audio "Digital Link" (see below), I suggest that you spend no more than $400 on the modification of a CD player.

CD players using the Burr-Brown PCM58P or PCM61P in its selected grade should also be considered. The **Onkyo DX-7500** ($700.00), **Pioneer PD-71** ($850.00), and **Sony CDP-608ESD** (900.00) are all excellent players and highly recommend. Each of these units has distinct advantages and disadvantages, which are discussed in full reviews in this issue. *[Again, successor models exist in the case of the Pioneer and the Sony, but the similarities are much greater than the differences.—Ed.]* These units, for a variety of technical reasons, may be more difficult to modify than the Philips CD-80. If you are contemplating having the units modified at a later date, you might want to talk to the modifier before purchasing the base unit. You should also look at the **CAL Icon** at $750.00. According to the manufacturer this unit is DC coupled and uses the PMI OP42 op amps. I have not examined this unit and cannot make a firm recommendation.

The recent release of separate digital-to-analog converter boxes is very good news. Complete reviews of the **Aragon D2A** and **PS Audio "Digital Link"** are in this issue. I can recommend both units. You should not choose the Aragon unit if you are using a passive preamp (see the full review). Both Aragon and PS Audio have made provisions in their designs to allow updating to new DACs and filter chips. The ability to update these boxes is an advantage over a high-end all-in-one CD player. (Aragon and PS Audio have not yet announced an upgrade to the Burr-Brown PCM63P or Analog Devices AD1862. I hope these companies will keep the promise to upgrade their products.) The major disadvantage of a decoder box is increased time-base jitter from the SPDIF interface.

I could not evaluate the **Proceed PDP** decoder box ($1295.00) because the manufacturer, Madrigal Audio Laboratories, refused to a send a schematic (or, indeed, a unit for review). Based on the limited information I have, I cannot see a justification for spending $300 more than the price of the Aragon or $496 more than the price of the PS Audio on this unit. As for the ergonomic problems of the **Meridian 208** (see the full review in this issue), they do not exist in the **Meridian 203** decoder box ($990.00). The unit was introduced too late to be reviewed along with the 208, but at least it is more realistically priced. The analog section is similar to that of the 208 and thus lacks the level of sophistication found in the Aragon and PS Audio units (the NE5534 op amp is used, for example). Also the 203 uses the obsolescent SAA7321 DACs. The only circumstance under which I would recommend this unit would be the availability of an upgrade path to the SAA7350 (or at a minimum the SAA7323). To my knowledge, such a program has not been announced.

At a higher price point, you might consider examining the new **Theta DS Pro Basic** decoder ($2000.00) and the **Wadia DigiMaster X-32** ($1995.00). Both decoders use a digital filter implemented on a general-purpose DSP chip set. Both represent good value given the sophistication of the technology used in these units. The **"Pre"** version of the Theta receives a full review in this issue. Wadia declined to send the unit for review. The Wadia unit uses a time-domain algorithm. This algorithm causes a high-frequency roll-off and results in a high level of out-of-band energy at the unit's output. Others' reviews of the Wadia have shown very nonlinear DACs. The principal advantages of the DigiMaster X-32 are the excellent jitter performance of the RockLok clock recovery circuit and the ability to reconfigure the glue logic and SPDIF decoder with PROM chips. Overall, I think the disadvantages are greater than the advantages.

Given the steady rate of improvement in CD player technology, I cannot recommend spending more than $2000 for any CD player or decoder.

Acknowledgement

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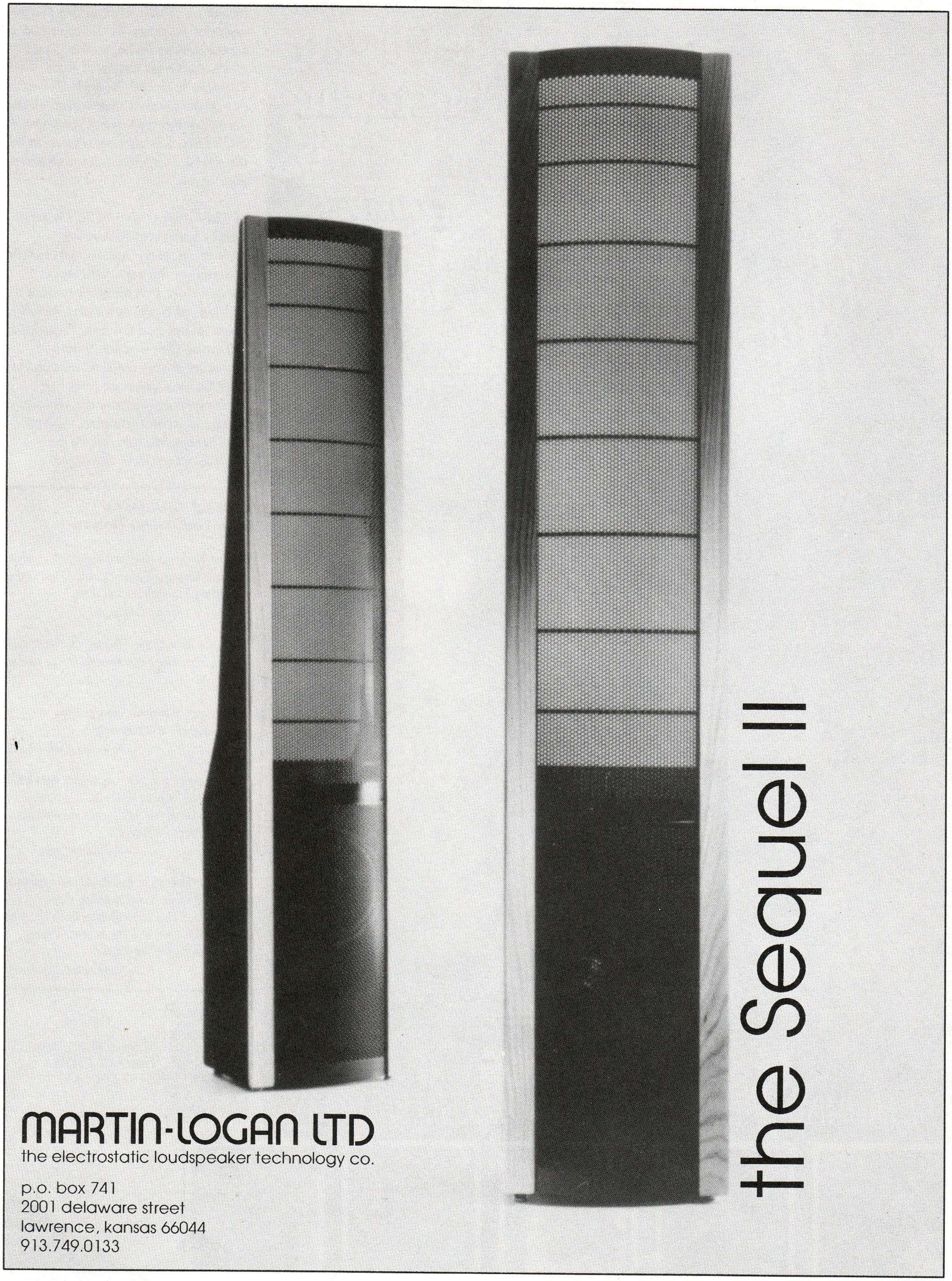
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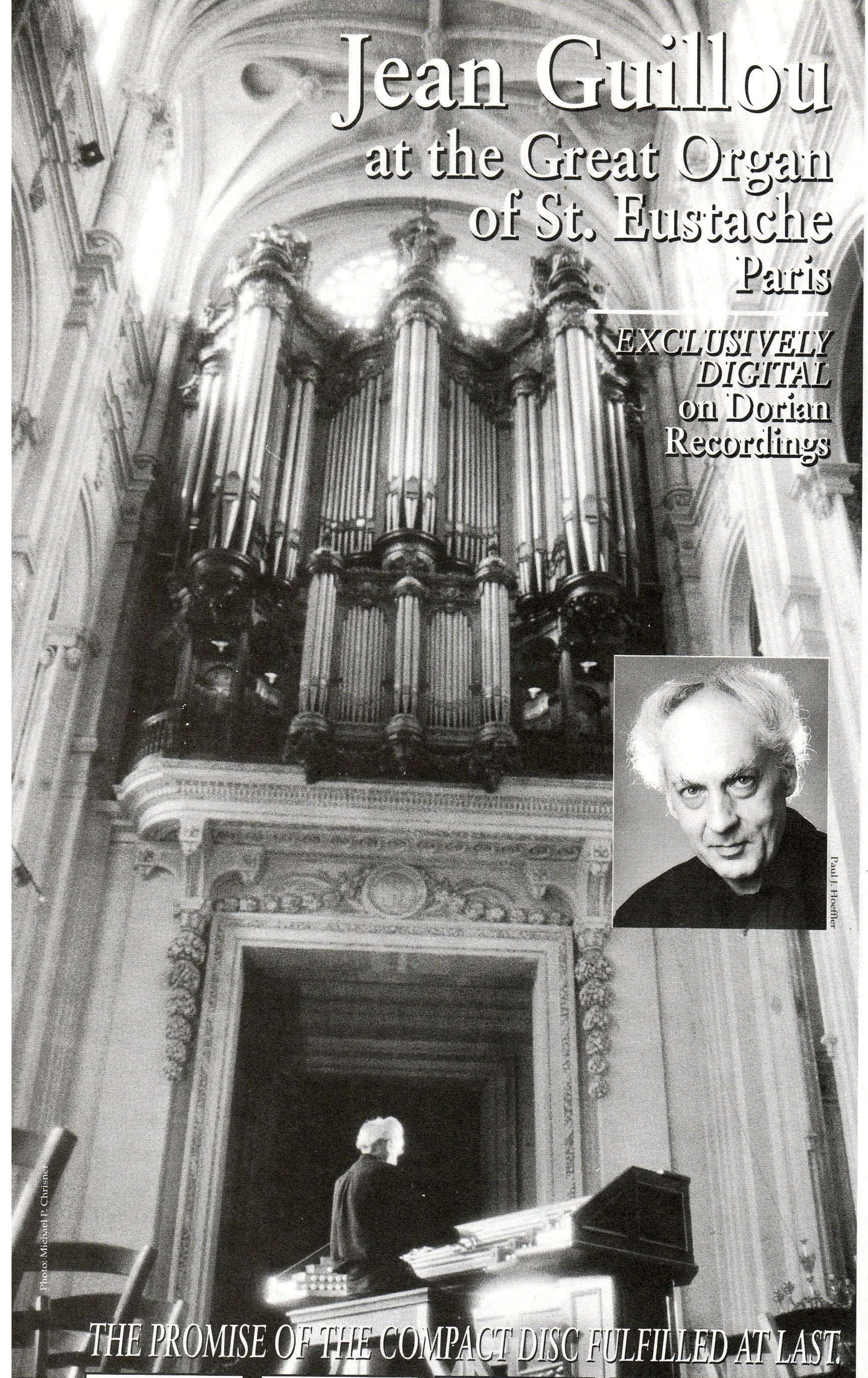

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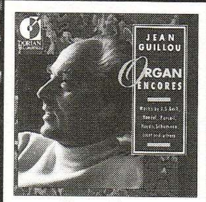

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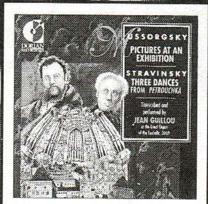
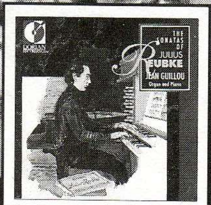
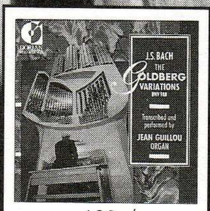
[DOR-90112]



Mussorgsky: Pictures at an Exhibition
Stravinsky: Petrouchka
[DOR-90117]



The Sonatas of Julius Reubke
(1834-1858)
[DOR-90106]



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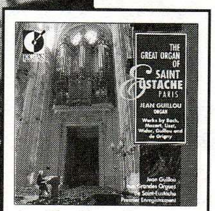
[DOR-90110]



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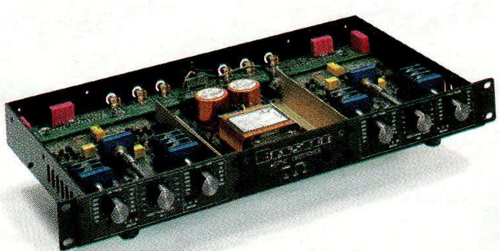

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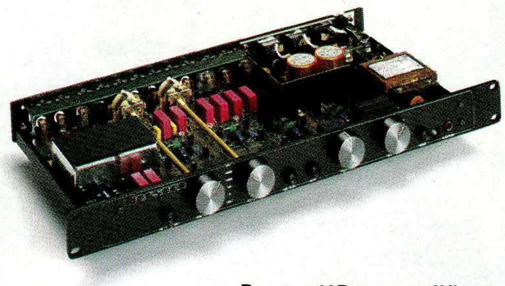

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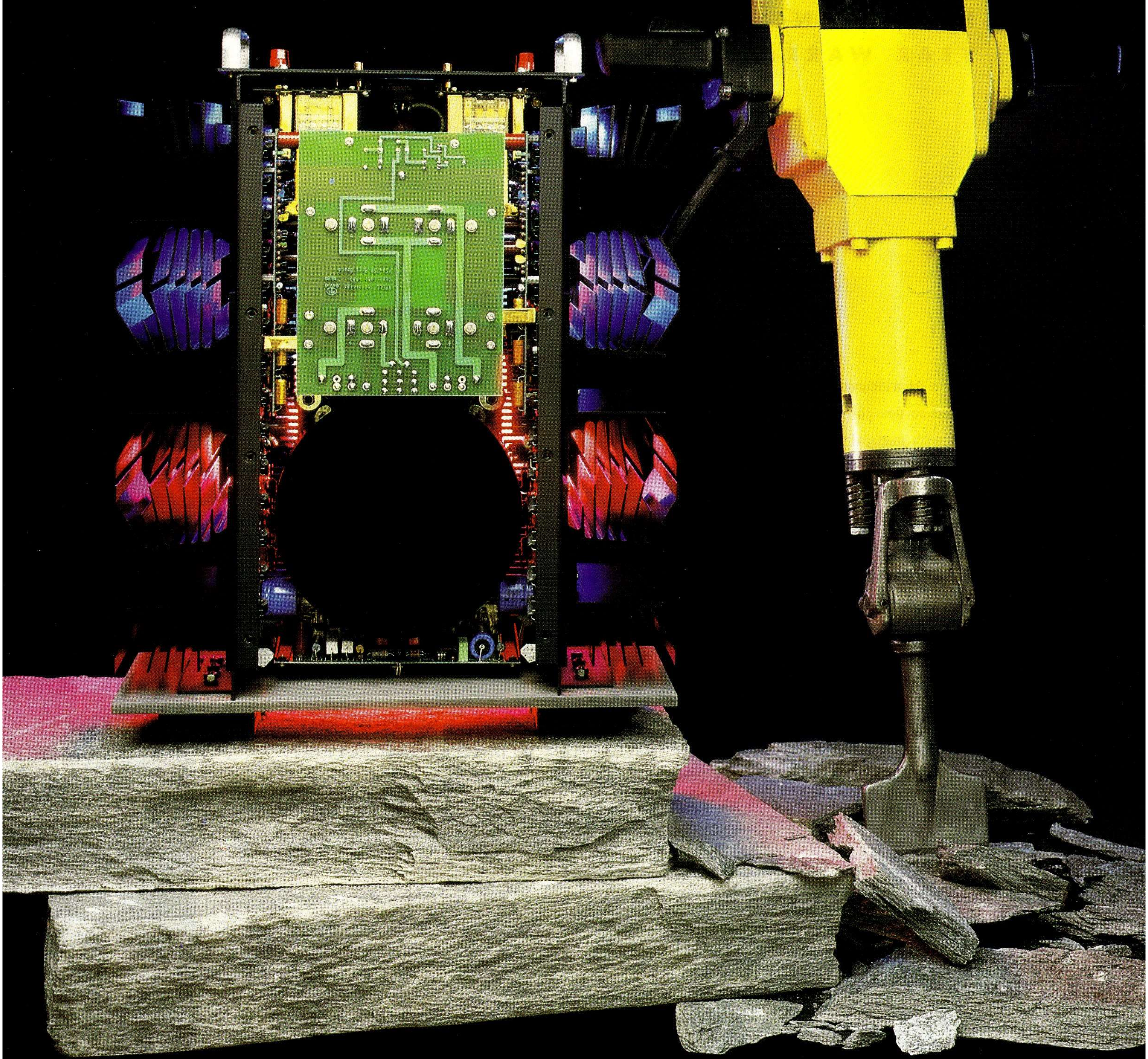

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(continued on page 40)

Table 1: Design Features of CD Players and Separate D/A Converters

	SPDIF Decoder [1]	Digital Filter	D/A Converter	Interpolation Rate	Sample and Hold Circuit	Analog Stages	I to V Converter
Aragon D2A	YM3623B + 2 addnl PLLs	Sony CXD1144	Burr-Brown PCM58P-J	8×	No	2 (discrete)	Voltage feedback
CAL Icon	NA	NPC SM5813	Burr-Brown PCM61P	8×	No	2 (integrated)	Voltage feedback
CAL Tercet Mk III	NA	NPC SM5813	Burr-Brown PCM61P	8×	No	2 (discrete)	
Carver TL-3220	NA	Yamaha YM3414	Burr-Brown PCM61P	8×	Yes	5 (integrated)	Voltage feedback
Denon DAP-2500	Yamaha YM3623B	Sony CXD1162	Burr-Brown PCM56P-J [3]	4×	Yes	2 (integrated)	Voltage feedback
Denon DCD-1560	NA	NPC SM5813	Burr-Brown PCM1701KP	8×	Yes	3 (integrated)	Voltage feedback
Harman/Krdn HD7500/7600	NA	Included in Bitstream DAC	NTT MN6471M	NA	NA	3 (discrete)	NA
JVC XL-Z1010TN	NA	Yamaha/JVC YM3414/K2	Burr-Brown PCM56P	8×	Yes	2 (integrated)	Voltage feedback
Kenwood DP-8010	NA	NPC/Kenwood SM5813/propr	Burr-Brown PCM58P	8×	No	3 (integrated)	Voltage feedback
Kinergetics KCD-40	NA	Sony CXD1144	Analog Devices AD1860N-K	8×	No	2 (composite)	Voltage feedback
Krell SBP-64X	Proprietary	4 Motorola 56001 [4]	Burr-Brown PCM64	64×	Yes	1 (discrete)	Voltage feedback
Magnavox CDB630	NA	Philips SAA7220P/B	Philips TDA1541A	4×	No	2 (integrated)	Voltage feedback
Meridian 208	NA	Included in Bitstream DAC	Philips SAA7321 [5]	NA	NA	2 (integrated)	NA
Onkyo Integra DX-7500	NA	Yamaha YM3414	Burr-Brown PCM58P-K	8×	No	Choice of 1 or 2 (integrated)	Voltage feedback
Philips CD-60	NA	Philips SAA7220P/B	Philips TDA1541A	4×	No	2 (integrated)	Voltage feedback
Philips CD-80	NA	Philips SAA7220P/B	Philips TDA1541A-S1	4×	No	2 (integrated)	Voltage feedback
Philips DAC960	Sony CXD1076 [6]	Philips SAA7220P/B	Philips TDA1541A-S1	4×	No	2 (integrated)	Voltage feedback
Pioneer Elite PD-71	NA	NPC SM5803	Burr-Brown PCM58P-K	8×	No	3 (2 composite & 1 integrated)	Voltage feedback
Proceed PCD	NA	NPC SM5813	Burr-Brown PCM61P	8×	No	2 (composite)	Voltage feedback
PS Audio "Digital Link"	Yamaha YM3623B	Yamaha YM3434	Burr-Brown PCM61P-K	8×	No	1 (composite)	Proprietary
Sansui AU-X911DG	YM3623B + 2nd PLL	Included in Bitstream DAC	NTT MN6471M	NA	NA	4 (integrated)	NA
Sony DAS-R1	Sony CXD1076 [6, 7]	Sony CXD1144 & CXD1329	Philips [8] TDA1541A-S1	8×	No	2 (1 integrated & 1 composite)	Voltage feedback
Sony CDP-X7ESD	NA	Sony CXD1244 & CXD8003	Burr-Brown PCM58P-S	8×	No	2 (integrated)	Voltage feedback
Sony CDP-608ESD	NA	Sony CXD1244 & CXD8003	Burr-Brown PCM58P-J	8×	No	2 (integrated)	Voltage feedback
Sony CDP-508ESD	NA	Sony CXD1244 & CXD8003	Burr-Brown PCM58P	8×	No	2 (integrated)	Voltage feedback
Theta DS Pro	YM3623B + 2 addnl PLLs	2 Motorola 56001	Analog Devices AD1860N-K	8×	No	2 (integrated)	Voltage feedback
Theta DS Pro Basic	YM3623B + 2 addnl PLLs	2 Motorola 56001	Analog Devices AD1860N-K	8×	No	2 (integrated)	Voltage feedback
Wadia 2000	Proprietary FLL (VCO/crystal)	4 AT&T DSP-16	Proprietary	64×	No	2 (hybrid)	Voltage feedback
Wadia Digi-Master X-64	Proprietary FLL (VCO/crystal)	4 AT&T DSP-16	Proprietary	64×	No	2 (hybrid)	Voltage feedback
Wadia Digi-Master X-32	Proprietary FLL (VCO/crystal)	2 AT&T DSP-16	Proprietary	32×	No	2 (integrated)	Voltage feedback
Yamaha CX-1000	Yamaha YM3623B [9]	Yamaha YM3414	Burr-Brown PCM56P-K [10]	8×	Yes	3 (integrated)	Voltage feedback
Yamaha CDX-1120Ti	NA	NPC SM5813	Burr-Brown [3] PCM58P-J [10]	8×	No	5 (integrated)	Voltage

[1] Applicable only to outboard converter boxes and digital preamplifiers.

[2] Total number of regulators in unit (outboard converters naturally require a smaller number than complete CD players or digital preamplifiers).

[3] Balanced operation with 2 DACs per channel.

[4] General-purpose DSP (digital signal processor) with custom software.

[5] A proprietary configuration allows one SAA7321 to be used in each channel.

(Blank space signifies unavailable data. See review section for successor models, if any.)

Filter Order	Output Coupling	Number of Transformers	Optical Couplers	Regulator Stages [2]	Dual Mono	Analog Sect Pwr Sup V	Price $
3rd order Bessel	Capacitor	1	No	11	Yes	±18	995.00
3rd order	Capacitor	1	No	5	No		750.00
3rd order	DC servo	2	Yes	23			1295.00
3rd order	Capacitor	1	No	5	No	± 8	529.00
3rd order Butterworth	DC servo	1	No	8	No	±15	1000.00
Passive (order unknown)	Capacitor	1	No	7	No	±12	650.00
3rd ord Btrw + 3rd ord pulse int	Capacitor	1	No	7	No	±12 (filter)	449.00 599.00
3rd order Butterworth	Capacitor	1	No	10	No	±10	700.00
3rd order Butterworth	Capacitor	1	No	7	No	±15	650.00
2nd order Butterworth	DC servo	2	No	15	No	±18	2295.00
1st order	DC servo	3	No	10 (not sure)			8950.00
3rd order Bessel	Capacitor	1	No	4	No	±15	399.95
2nd ord Btrw + 3rd ord pulse int	DC servo		No	5	No	+ 5 (pulse int) ±15 (filter)	2950.00
Choice of 1st or 3rd order	DC servo	2	Yes	7	No	±15	700.00
3rd order Bessel	Capacitor	1	No	4	No	±15	429.95
3rd order Bessel	DC servo	1	No	12	No	±15	799.95
3rd order Bessel	Capacitor	3	Yes	7	No	±15	999.00
3rd order Butterworth	Capacitor	1	No	7	No	±15	850.00
	DC servo	1	No	11			1650.00
1st order	Direct (with DC offset canc pot)	1	No	6	Yes	±15	799.00
1st order + 3rd order pulse int	Capacitor	1	No	7	No	±15	1100.00
3rd order Butterworth	Capacitor	2	No	7	No	±15	8000.00 (with CDP-R1)
3rd order	Capacitor	2	No	10	No	±15	2000.00
3rd order Butterworth	Capacitor	1	No	5	No	±12	900.00
3rd order Butterworth	Capacitor	1	No	3	No	± 5	550.00
1st order	DC servo	4	No	13	Yes	±15	3500.00
1st order	DC servo	2	No	7	No	±15	2000.00
RF filtering only	Direct (with DC offset canc pot)	2	No		Yes	±15	7995.00
RF filtering only	Direct (with DC offset canc pot)	2	No		Yes	±15	4995.00
RF filtering only	Direct (with DC offset canc pot)	1	No		Yes	± 8	1995.00
1st order	DC servo + capacitor	2	No	18	Yes	±12	1199.00
Choice of 1st or 3rd order Btrw	Capacitor	2	No	5	No	±12	1199.00

[6] Only the PLL phase detector is integrated on this chip.

[7] Clock signal is internally generated when used with Sony CDP-R1 transport.

[8] Staggered configuration with 2 DACs per channel provides 2x increase in conversion rate.

[9] External adaptive PLL filter is used. Crystal oscillator is stopped when phase lock is achieved.

[10] Bit shifting circuit is used after the DACs.

Table 2: Design Features of Modified CD Players

	Models Modified	D/A Converter Replaced	Analog Stages	I to V Converter	Filter Order
Euphonic Technology Mk II Signature Arpeggio	Magnavox CDB-650 & most other Philips-based players	Yes, with Philips TDA1541A-S1 + separate transformer in DAC power supply	2 (integrated)	Voltage feedback	3rd order Bessel
MSB Technology	Magnavox				
Silver 630	CDB-630	No	2 (composite)	Transimpedance	1st order
Gold	CDB-582	Yes (TDA1541A-S1)	2 (composite)	Transimpedance	1st order
Paul McGowan Designs	Any player with >±12 V supply rails	No	1 (composite)	Proprietary	1st order
Precision Audio DVIC	Any player with PCM58 or TDA1541 DACs and >±12 V supply rails, as long as mod fits chassis	No	1 (discrete)	Transimpedance	2nd order (Q = 0.5)

Table 3: Comparison of 8 Times Interpolating Digital Filters

	NPC SM5803	NPC SM5813	Sony CXD1144	Sony CXD1244	Yamaha YM3414 YM3434
Passband Ripple	±0.00005 dB	±0.00005 dB	±0.000005 dB	±0.00001 dB	±0.00005 dB
Passband Frequency Response	±0.00005 dB	±0.00005 dB	+0, –0.003 dB	±0.00001 dB	±0.00005 dB
Stopband Attenuation					
24.1–150 kHz	–115 dB	–115 dB	–120 dB	–100 dB	–100 dB
150 - 180 kHz	–115 dB	–115 dB	– 70 dB	–100 dB	–100 dB
Noise Shaper	Selectable	No	No	Selectable	No
Digital De-emphasis	Yes	No	No	Yes	No
DAC DC Offset	No	No	No	Selectable	No
Number of DSP Processors	1	1	2	1	1
FIR Tap Length	199	199	293	Not available from manufacturer	287
Coefficient Word Length	22 bits	22 bits	28 bits	Not available from manufacturer	19 bits
Data Path Length	20 bits	20 bits	22 bits	Not available from manufacturer	18 bits
Accumulator Size	25 bits	25 bits	32 bits	45 bits	18 bits

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(continued from page 37)

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(Blank space signifies unavailable data.)

Output Coupling	Replace Filter & De-emphasis Components	Dual Mono	Output Muting Function	Price of New Modified Player $	Price to Modify Your Player $
Capacitor (film)	Yes, with Philips SAA7220P/B	No			Varies
				1595.00	
				1200.00	
DC servo	Yes		Removed	1695.00	
DC servo	Yes		Removed	2350.00	
Capacitor	Yes	No	Removed		300.00
DC servo	Yes	No	Relay	1300.00 (Philips CD-80)	450.00 to 550.00 (depending on player)

Table 4: Comparison of ICs Suitable for Use in CD Players

VOLTAGE FEEDBACK

	Settling (ns) 0.1%	0.01%	Noise ($nV/\sqrt{Hz}$ @ 1 kHz)	Slew Rate (V/µs)	Small Signal Bandwidth (MHz)	Output Cur for ±10 V (mA)	Input Bias Current (µA/nA/pA)	Price ($ per chip in lots of 100)
Analog Devices								
AD841	90	110	13	300	40	50	3.5 µ	6.00
AD845	250	310	25	100	12.8	20	500 p	3.25
Burr-Brown								
OPA627	450*	550*	5.6	135*	16	45	20 p	7.50
OPA602	600	1000	13	35	6.5	20	10 p	4.50
OPA606	1000	2100	14	35	13	10	10 p	3.85
National								
LM6361	120	Not specified	15†	300	35	7	3 µ	1.91
LF400	200	365	23	27	16	20	200 p	3.95
LM833	Not specified	Not specified	4.5	7	15	Not specified	500 n	1.05 (dual)
PMI								
OP42	450	1000	13	50	10	10	80 p	2.25
Signetics								
NE5532	Not specified	Not specified	5	9	10	20	200 n	1.44 (dual)
NE5534	Not specified	Not specified	4	13‡/6	10	20	200 n	1.00

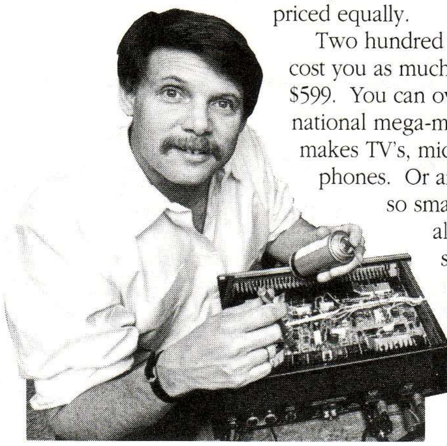
*@ $A_{VCL} = -1$ †@ 10 kHz ‡@ $A_{VCL} = 3$

TRANSIMPEDANCE (CURRENT FEEDBACK)

	Settling (ns) 0.1%	0.01%	Slew Rate (V/µs)	Small Signal Bandwidth (MHz)	Output Cur for ±10 V (mA)	Price ($ per chip in lots of 100)
Analog Devices						
AD846	80	110	450	46	20	6.25
Burr-Brown						
OPA603	50	Not specified	1000	55	80	4.95
Elantec						
EL2020	90	Not specified	500	50	50	3.30
Harris						
HA5004	50	Not specified	1200	100	95	5.87
PMI OP160	120	155	1000	90	20	4.50

HOW MUCH SHOULD A GOOD AMPLIFIER COST?

Reflections on the esoteric myths and economic realities of power amplifier design, by Bob Carver.



Thumb through *Audio's* Annual Equipment Directory and you'll see vivid proof that all power amplifiers are neither created equal nor priced equally.

Two hundred watts per channel can cost you as much as $8,400 or as little as $599. You can own an amp from a multinational mega-manufacturer who also makes TV's, microwaves and cellular phones. Or an amp from a company so small that the designer is also the assembler and shipping clerk.

Can it be that amplifiers are sonically equal? Some seem to have muscular power reserves far beyond their FTC-rated output. Others sound great until they're challenged by a dynamic passage and then sound like a Buick hitting a row of garbage cans. Some are (to indulge in audiophile jargon) so "fluid" that you practically need a drop cloth under them. Others seem to sound harsh, "metallic" and brittle at any output level.

A casual comparison of perceived sound quality versus price tags may lead to an erroneous conclusion: that an amplifier must be *expensive* to sound good.

The truth is a bit more complicated: Cosmetic glitz aside, an amplifier's cost is primarily determined by its power supply.[1] In other words, within reason, you generally do get what you pay for when you buy a conventional amp design. But the key word here is "conventional."

My decidedly *un*-conventional Magnetic Field Power Supply is capable of outperforming conventional power supplies of the same size. Result: A significantly better power amplifier value for you.

Let me explain.

NO MAGIC. JUST FOUR CRITICAL QUANTITATIVE FACTORS.

When I fervently state that "*the sound of an amplifier need not be related to its price,*" you might think we're veering off into the land of Snake Oil and Gimmicks. Quite the contrary.

I and other members of the scientific audio community know that just four factors determine the sonic characteristics of an amplifier:

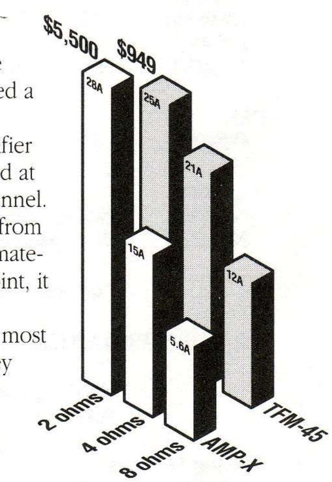
1. Current output
2. Voltage output
3. Power output
4. Transfer function *as evidenced by the interrelationship of frequency response and output impedance.*

These factors transcend the usual trivial debates over tubes vs. solid state, MOS-FETs vs. bi-polar, Class A vs. AB, silver Leitz wiring vs. copper, gold-plated front panels, WonderCaps and my favorite: hand-ground-open transistors filled with a proprietary crystalline substance that stops ringing (honest, I'm not kidding!). An amp can have any combination of these entertaining variables (plus special bricks stacked on top) and yes, sound wonderful...provided it ALSO has high current, voltage and power output and the correct output impedance.

Thus the Four Factors explain why expensive amplifiers generally sound better than cheap amplifiers. But also why that doesn't necessarily have to be the case.

FACTORS 1-3: THE POWER SUPPLY BEHIND THE SOUND

An amplifier's power supply produces current and voltage. A preponderance of one without the other is meaningless.[2] To maximize SIMULTANEOUS current and voltage output using traditional design approaches costs serious money. For example, we recently tested a competitor's $2,000 amplifier that was rated at 20 watts/channel. Believe me, from a parts and materials standpoint, it was worth $2,000, with most of that money being spent on an amazingly rugged power supply. Another more extreme example is my own ultra-conventional Silver Seven Tube amplifier design. Its "money-is-no-object" power supply helps set the price of a pair of S-7's at around $20,000.00.



Absolute maximum output
CURRENT
each channel

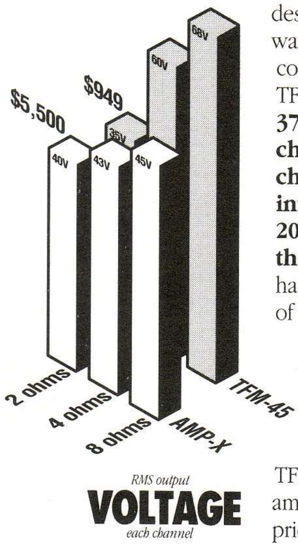
Now, since it is universally agreed among amplifier designers that current/voltage/power output directly affects the sound of an amplifier, and since good traditional power supplies are costly, price and sonic quality ARE often closely related.

But what if there was a way around the economic constraints of conventional, inefficient power supplies? What if there was a power supply that could deliver awesome simultaneous current and voltage into real-world speaker impedances without shocking your pocketbook?

TFM-45; 375 watts RMS/ch. into 8Ω 20-20kHz with no more than 0.5% THD ($949 sugg. retail.)and TFM-15; 100 watts RMS/ch. into 8Ω 20-20kHz with no more than 0.1% THD ($399 suggested retail).

That's just what my patented Magnetic Field Power Supply does. Without gimmicks, mysticism or loss of bass response. Simply put, a Magnetic Field Power Supply uses progressively more of each line voltage swing as amplifier power demand increases. It's just plain more efficient. How and why this works is explained in our new White Paper called *"The Magnetic Field Story Parts I, II & III"* which you can get free by calling 1-800-443-CAVR.

Right now, let's consider the tangible benefits. The series of comparison charts in this ad shows how my Magnetic Field Power Supply successfully challenges the previously hard-and-fast rule that high-performance power supplies must be expensive. Amp X is a highly-respected solid state design rated at 200 watts into 8 ohms. It cost $5,500. My TFM-45 is rated at **375 watts per channel both channels driven into 8 ohms 20-20KHz with less than 0.1% THD.** It has a suggested retail of $949.



Even more impressive is this same sort of comparison chart with the TFM-45 vs. other amplifiers in its own price range. In deference to how utterly we trounce similarly-priced, conventional competition, we've confined those charts to our new White Paper.

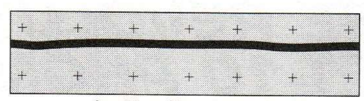
To summarize: Magnetic Field Power Supply technology allows reasonably-priced power amplifier designs to deliver simultaneous current and voltage levels previously only found in extremely expensive "esoteric" designs. Or to look at it another way, in a given price range (say $900-$1,000), Carver simply gives you far more for your money.

FACTOR 4: TRANSFER FUNCTION

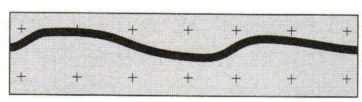
Consider two hypothetical amplifiers with identical power supplies. Same power rating; same gain, etc. Yet they still sound different when powering identical speakers through identical cables.

Why? A fourth quantifiable factor is at work. One that, unlike power supply output, is totally independent of economic constraints.

I've left Factor 4 (transfer function/frequency response/damping) until last intentionally. Because until an amplifier can deliver sufficient power with simultaneous current and voltage (Factors 1-3), transfer function is immaterial.



Amplifier with resistor test load



Same amplifier connected to cables and loudspeaker.

Frankly, I'm guilty of not making this fully clear in the past. Some readers may have gotten the impression that by magically adjusting some arcane parameter called transfer function, one could somehow cause a cheap amp to sound like an expensive one. Nothing could be further from the truth. If there's no guts (power supply), there's no glory (optimized transfer function).

By transfer function, I mean the effect an amplifier's output impedance has on real world frequency response. I don't mean the flat, "DC to light" Rated Full Power Bandwidth found in column 11 of *Audio's* Equipment Directory, which is measured using a resistor as a load. Rather, I'm referring to the frequency response curve that occurs when an amplifier and speaker cables interact with a specific speaker.

As distinctive as a fingerprint, this curve determines the "sound" of each amplifier design. Its warmth or harshness. The quality of the bass. The definition of its upper registers. Even the configuration of the stereo "sound stage" it can create.

My engineering department and I are capable of making one amplifier design sound like another amplifier design to within 99 parts out of 100 (a null of 40dB). For example, we've used Transfer Function Calibration to closely emulate the sonic characteristics of my reference Silver Seven in our TFM-45 and TFM-42 solid state designs. In other cases we've used the process to simply adjust the sound of an amplifier to have pleasant but unique sonic characteristics: in general, a warm "tube" sound with rich, rolling bass and soft yet detailed treble (such as our TFM-22/25 , S-7t and TFM-15). Either way, we use painstaking measurement and adjustment processes to finetune output impedance/frequency response. Not magic.

And, needless to say, we start with highly capable power amplifier designs before the Transfer Function Modification process.

ARE YOU INTRIGUED...OR THREATENED?

My Transfer Function Calibrated power amplifiers have suggested retail prices of from $399 to $1,000. That I even dare to suggest they can sound as good as designs in the $2,000 to $6,000 price range has not endeared me with some audiophiles or underground magazine writers.

That's a real shame, because I have absolutely nothing but respect for well-made, high-ticket conventional amplifiers. Like Rolexes and Lamborghini's, they are a joy to own if you can afford them. But just as a Rolex doesn't tell time any better than the inexpensive watch I'm wearing right now, good sound does not necessarily have to be costly.

If this concept intrigues you, please visit a Carver dealer soon. Bring demo material you're familiar with and be willing to do some critical listening. Compare my designs to competition costing about the same amount as well as to more expensive models.

Your ears alone should be the final arbiter. I feel confident that you will join the tens of thousands of audiophiles who have gotten the best possible value by owning Carver.

Bob Carver

Bob Carver, President

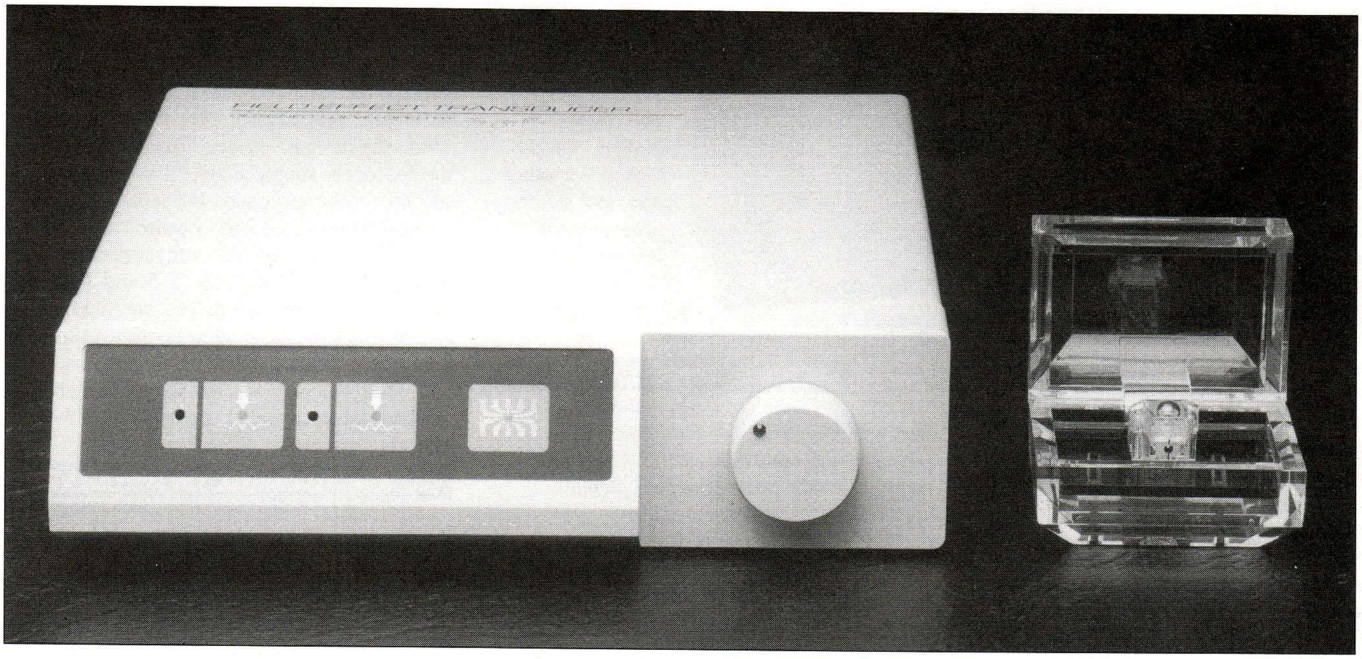
CARVER

CARVER CORP., LYNNWOOD, WA, U.S.A. 1-800-443-CAVR
Distributed in Canada by Evolution Audio Inc. 1-(416) 847-8888

[1] *My definition of cosmetic glitz is any part of an amplifier whose sole audio contribution is to cause one's friends to go, "Ooooo!!" when they see one's new purchase. My own Silver Seven amplifier's hand-rubbed piano lacquer and solid granite surfaces meet this definition.*

[2] *Since power (watts) equals voltage times current, the same wattage can represent significantly different combinations of voltage and current — and thus very different performance into the same load.*

BREAKING ALL BARRIERS...



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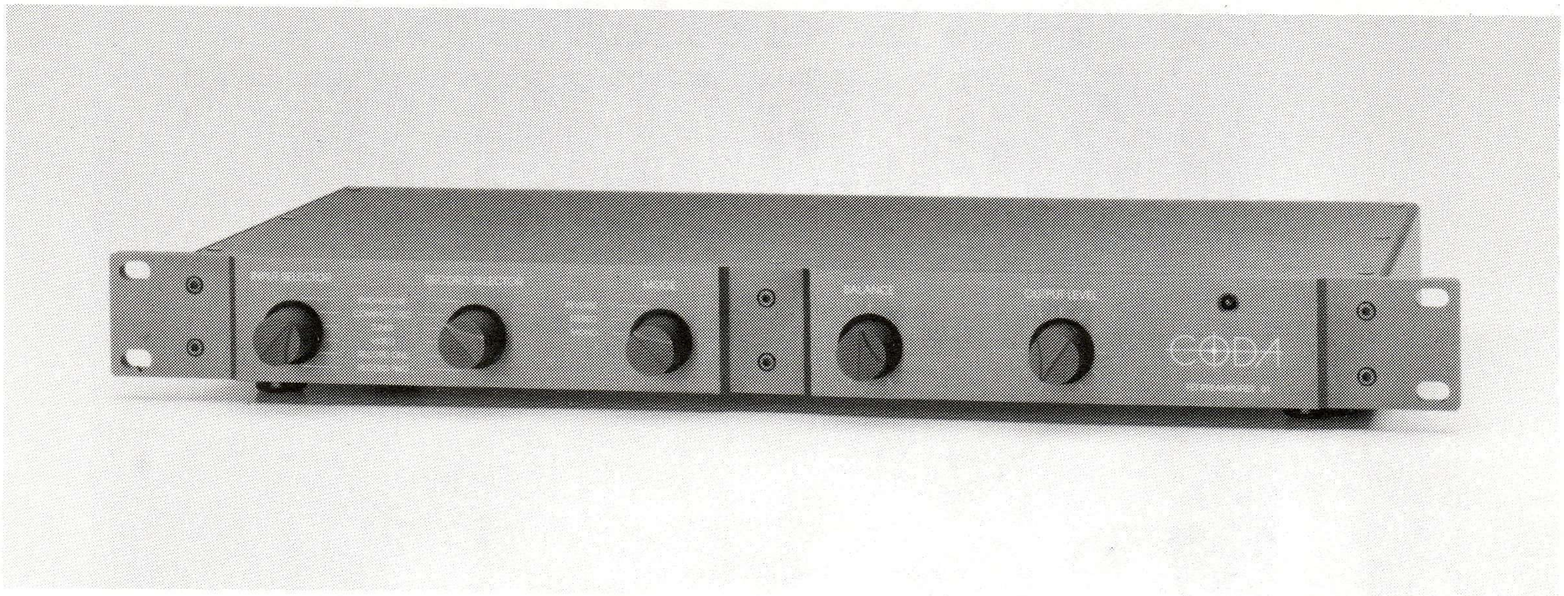

co•da \\'kōdə\\ *n* -s [**It**, lit., tail, fr. L *coda*, *cauda*] : a final or concluding portion of a musical or dramatic work ; *usu* : a portion or scene that rounds off or integrates preceding themes or ideas

Coda Technologies is a new expression of art and science at a level of refinement and quality that is rare in the audio industry. The company is the creation of a small group of dedicated engineering audiophiles whose goal is to utilize the best techniques and materials that are available and appropriate to a given design. Providing a foundation for these efforts are over two decades of experience in pursuing a sonic and aesthetic ideal. Coda Technologies products are limited production items, which are available at carefully chosen audio dealers, allowing the company to interface more completely with the audiophile user than would be possible for a company with a more "Mass Market" orientation. As with any audiophile product the proof is in the listening experience. We encourage you to audition the Coda Technologies FET PREAMPLIFIER 01 and realize its superior quality for yourself. For more information about Coda Technologies or its products please call or write us.

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Current CD Players and D/A Processors, New and Not So New, Multibit and One-Bit

By Peter Aczel
Editor and Publisher

Inspired by David Rich's formidable treatise, your Editor tests a bumper crop of hardware on the lab bench, in the listening room, and some through the ABX Comparator (if you'll pardon the expression).

My favorite cliché, "the plot thickens," is quite safely applicable here, since just about every manufacturer capable of mounting a printed circuit board on a chassis has come up with CD playback hardware of some sort, more often of several sorts, and rival design philosophies are proliferating at all price levels, not just the high end. Armed with the technical insights provided by David Rich, I managed to keep my head above the water in the lab, but boy, there are more difficult choices in this sector of audio today than on the menu of a good Chinese restaurant. Let us, therefore, try to sort out the relevant issues and criteria.

How they differ, how they don't.

All CD playback equipment currently or recently manufactured can be expected to have almost dead flat frequency response, negligible de-emphasis error, stupendous stereo channel separation, virtually zero phase difference between channels, extremely low THD at reference level, insignificant wow and flutter, and nearly always noninverting impulse response. The elaborately presented curves for these measurements in the typical magazine review are mere gingerbread in my opinion (look, Ma, I have an Audio Precision!); my trusty Hewlett-Packard 3580A has a guaranteed amplitude accuracy of ±0.3 dB and a visual resolution of about 0.1 dB, leaving me little or nothing to say about even smaller differences that are meaningless in any event. At the same time, there are very real though rarely important differences in noise floor, dynamic range, 0 dB square wave clipping, RF output, possibly IM distortion and, most significantly, low-level gain linearity and monotonicity. These are worth discussing, but here again I consider a 0.3 dB deviation from theoretical perfection, or a 0.3 dB superiority in device A versus device B, to be quite insignificant and not really what the enlightened audiophile needs to know.

What he needs to know, rather, is whether or not the unit is as well engineered—with regard to basic circuit concept, quality of parts, mechanical construction, control facilities, and ergonomics—as he has the right to expect for the money he is paying. Significant differences exist in those respects. What about the sound? I'll be coming to that shortly.

Low-level linearity: not so simple.

When the deviation from theoretically perfect gain linearity with a dithered test signal at –70 dB, –80 dB, and –90 dB is small enough to be approaching the accuracy and resolution limits of my instrumentation (see above), I call the gain linearity essentially perfect, although it can happen that the spectrum of the signal still shows some harmonic blips above the "grass" of the noise floor. Usually, when the amplitude deviation is a fraction of a dB, no such blips are visible. Quite a few units turned out to be that good, even at -90 dB, but a number of those that showed some nonlinearity led me to a totally unexpected conclusion.

I now believe, very firmly, that trimmer potentiometers should be left out of designs using the J-grade and K-grade DACs from Burr-Brown and Analog Devices. Those who use these DACs as is, exactly as they come from the factory, get essentially perfect linearity (e.g., Sony and PS Audio), whereas those who add the trim pots (e.g., Pioneer Elite and Theta) end up with far from negligible errors in actual production samples, correctable of course on a suitably equipped lab bench but not by the consumer in his home. In fact, an untouched J-grade appears to be more linear than a typical trimmed-in K-grade. The lowest grades, on the other hand, should always be used with trim pots to bring their linearity up to a respectable par, but production tolerances are even sloppier in the lower-priced CD players, so that the end result is generally unpredictable.

The irony of it all is that the beautifully plotted low-level linearity curves in all those highly detailed test reports are merely an indication of how carefully Norma tickled and sealed the trim pots on the production line, and how gently José handled the carton in the warehouse, but not of the inherent linearity of the design itself. They are QC curves, not engineering test curves. I have learned to take them *cum grano salis.* I am aware that Stanley Lipshitz and John Vanderkooy already flagged this trimmer pitfall in their March 1988 paper, but at the time they did their research the alternative of almost perfectly linear DACs right out of the package did not yet exist.

Double-blind, matched-level listening tests.

The point of any or all of the above is of course the obtainment of the best possible sound. I carefully listened at some length to every piece of equipment reviewed here, and a number of my associates also listened to most (but not all) of them. In a good many instances we had two of them level-matched within 0.1 to 0.15 dB of each other (meaning all four channels) and connected to the ABX Double-Blind Comparator. It would have been impossible, of course, to ABX each unit against each of the others; the permutations and combinations would have been staggering. We did enough double-blind listening, however, to come to a tentative conclusion that will not please the tweaks and cultists, or the major-brand lobbyists for that matter. There are simply no reliably identifiable differences in sound between any two units in this group, regardless of price, reputation or measured performance.

Those who can handle that without freaking out and showing me the door (as in "Get lost, Julian!") will want to know the details. The listening setup consisted of a pair of Quad ESL-63 USA Monitors driven by a Boulder 500AE power amplifier, which was fed from the balanced outputs of a Boulder MS preamplifier. In some of the listening tests a Velodyne ULD-15 Series II subwoofer was added to the Quads. The room was the one analyzed in Bill Rasnake's article in Issue No. 13 All of the participants in the tests were experienced audiophiles with no hearing impairments. The typical test between a given A and given B consisted of 16 trials, i.e., 16 successive randomized X's, with unlimited time available for each X and backtracking/rechecking permitted at all times. Some of the tests were stopped at 12 trials. The participants listened one by one, not in groups; the typical time they needed for 16 trials was about 60 to 75 minutes, some of which went into synchronizing the two identical discs (except when only one disc was needed to test a CD player straight through versus an outboard D/A processor). The music was quite varied—symphony orchestra, piano, soprano voice, string quartet, jazz, rock—but of course there was no way to include each listener's favorite test CD. Statistically, 12 right answers out of 16 and 10 out of 12 are needed for 95% confidence that no lucky guessing is involved. Nobody came even close to that; in fact nobody exceeded a fifty-fifty score by a statististically meaningful margin. And I must add that a Golden Ear who tells me that such and such "blows away" thus and so—"night and day" and "not in the same league" are in a similar idiom—had better get 16 right out of 16 or I consider his credibility blown. For 12 out of 16, the words should be "maybe a little better" or "I can just barely hear it."

Now for for the qualifications and reservations. It is possible that there exists a piece of music that would have revealed small differences better than anything used in these tests. It is also possible that, currently used systems of A/D encoding being less than theoretically perfect, some future encoder will reveal such small differences regardless of the music used. Remember, some of these differences were measurable. Finally, it is possible that there are some amazing Golden Ears out there who would have scored higher in the tests than any of us. But, please, don't talk to me about tremendous differences in sound quality between CD players of current or recent manufacture because I'll laugh in your face. (And don't hassle me about the ABX method because I'll instantly agree to, say, switching by hand-plugging and unplugging without relays, or any kind of time frame—such as one trial a day for 16 days—or any other method, as long as you match the levels and are listening to an unidentified sound when you give your answer. What I want the hasslers to explain to me is why every Golden Ear I or my associates have tested suddenly turns to tin when the levels are matched and the brand names are withheld. If you don't have an answer to that, please don't bore me with ecclesiastical arguments about side issues.)

Aragon D2A

Mondial Designs Limited, 2 Elm Street, Ardsley, NY 10502. D2A D/A converter, $995.00. Tested sample on loan from owner.

This is a gorgeous piece of equipment, somewhat handicapped by a wrongheaded decision regarding the analog output interface. The output stage is of the common emitter configuration, with a very high output impedance, unsuitable for driving a low-impedance volume control (as in a passive preamp or input-attenuated power amp) and also a poor match for high-capacitance interconnect cables. A properly designed active preamp will wash out these incompatibilities—but what if you have, or want, one of those trendy passive preamps, or no preamp? The fully discrete analog section (by itself a noble idea) has the further small boo-boo that the differential pairs are biased by a resistor which causes the bias current to vary in the presence of a common-mode signal, resulting in increased distortion. A minor engineering simplism.

Other than that, the D2A is a most attractive buy. You take off the cover and you can see where the money went; the parts and construction are of near-military quality, and the digital circuitry is highly sophisticated, with no chintzy solutions apparent in the details. The digital filter is the very expensive and complex Sony CXD1144BP. I don't even quite understand how Mondial does it for $995. The DAC is the J grade of the 18-bit Burr-Brown PCM58P, which is good enough in my book, but yes, there are trim pots, and no, they weren't perfectly adjusted in my sample. I measured a small amount of low-level nonlinearity that wouldn't have been there in my opinion with just the naked, factory-trimmed J grades (see above). On the other hand, the D2A is loaded with quality features, such as two coaxial and one optical digital inputs (no tweaky "optophobia" here!), a digital absolute-phase inversion switch (wow!), a pre-emphasis indicator light (an almost extinct but still relevant convenience), and more.

Summa summarum, this could have been the perfect D/A processor for the audiophile who knows value when he sees it, if only a few little engineering decisions had gone the other way. Even so, any owner of the Aragon D2A who is unaffected by those decisions can be justly proud of it. It's a good machine.

Carver TL-3220

Carver Corporation, P.O. Box 1237, Lynnwood, WA 98046. TL-3220 compact disc player with remote control, $529.00. Tested sample on loan from manufacturer.

Here we are at the bottom of the price range covered in this survey, meaning parts and construction typical of midpriced made-in-Japan audio components, a few nice little Bob Carver touches, some inevitable engineering tradeoffs, the absence of certain features—you get the picture. Nonetheless, amigos, the sound of the Carver TL-3220 was statistically indistinguishable in double-blind, matched-level listening from that of the $850 Pioneer Elite PD-71 (to name just one example), which is a thoroughly audiophile-oriented product. Sorry about that.

This is also an 18-bit, 8 times oversampling design, but the DAC is the lowest-grade Burr-Brown PCM61P, which works best with a little trimming. The trim pots appeared to be reasonably well aligned, since the low-level gain linearity and harmonic distortion were quite acceptable though not outstanding. For reasons unknown, a sample-and-hold circuit is included. Other eyebrow raisers are the very low ±8 V supply rails in the analog stage, low-voltage op amps of unfamiliar designation, electrolytic capacitors in the output signal path—I could go on, but then I have no solid proof that the avoidance of such audiophile hang-ups has any audible effect.

One of the unique Bob Carver touches is the—you guessed it—Digital Time Lens, the flamboyantly named signal processor circuit designed to make early CDs sound more like LPs. It does that by softening the upper midrange and increasing front-to-back depth with some L – R tweaking. One thing is certain: when you press the DTL button, the Carver TL-3220 no longer sounds like the Pioneer or any other CD player. I have very little use for it, but it may conceivably be somebody else's main reason for buying the Carver. I do wish, however, that the DTL circuitry were bypassed instead of merely deactivated when the button is in the out position. On the other hand, I rather like Bob's highly personal ergonomic layout of the front-panel control buttons, quite different from standard Akihabara issue.

My severest criticism of the TL-3220 is that it lacks a digital output, either coaxial or optical, and that neither the front panel nor the remote control has index search buttons. A CD like the Richard Strauss *Eine Alpensinfonie* (Telarc CD-80211) has just one track but 22 indexes. The TL-3220 is incapable of selecting one of the latter; it can only be fast-forwarded to the approximate location, and even that takes forever. Yeah, I know, Bob—that old Buddy Holly CD transfer you take with you everywhere doesn't have that problem.

Euphonic Technology Mk II Signature

Euphonic Technology, 19 Danbury Road, Ridgefield, CT 06877. Mark II Signature compact disc player with remote control, $1595.00. Tested sample updated from the ET650PX reviewed in Issue No. 11.

Michael Goldfield is one of the most highly endorsed Philips modifiers in the delirious little world of high-end audio. He does a beautiful job and then charges far too much, for the simple reason that you can't make a living charging a fair price for Philips mods. Therefore he absolutely needs a cult following—and he gets it. His mods just sound more "musical" than anything else you can buy, the true believers will tell you. (To paraphrase Samuel Johnson outrageously, musicality is the last refuge of a tweak.)

Actually, the Mk II Signature differs from the 1987-vintage ET650PX reviewed in Issue No. 11 in only five ways that I can discern: the DAC is now the TDA1541A S1 "Golden Crown" chip, not the plain-vanilla TDA1541; the digital filter is similarly upgraded to the SAA7220P/B; the DAC power supplies are new; the headphones output is "disabled for sonic benefit" (tweak tweak hooray!); and the price is up $600. And for your $1595 you still get the antediluvian and relatively insubstantial Magnavox CDB650 chassis (reinforced with the Euphonic Technology DPS-1 stabilizer, to be sure) and the Motorola MC34082 analog output chip, wholesale cost $1.00. (Mike is very secretive about these chips, having obliterated all identification on them and refusing to reveal their provenance, but circumstantial evidence and educated guessing point insistently to the Motorola. "If this be error and upon me prov'd," blame it on the secrecy.)

Now consider the Philips CD-80 reviewed below. Same DAC. Same digital filter. Just as good, or better, power supply regulation. The greatly improved NE5534 decompensated op amp in the analog stage (rather than the Magnavox's NE5532 which Mike disliked so much that he had to replace it with the Motorola). Plus a heavy-duty cast-alloy chassis with superior moving parts and better ergonomics. All that for $799.95, barely half the price of the Mk II Signature. See what I mean about the Philips mod business? To tip the scales even more cruelly, the gain linearity at –90 dB in one channel of my Signature sample was off by more than 3/4 LSB, which is out of spec for the Golden Crown DAC, and the noise floor of the Signature in both channels was about 5 dB higher than that of the CD-80. The DAC glitch is probably just a QC slipup, but the noise is not; the Motorola chip can be assumed to be the culprit there.

None of this makes Michael Goldfield's handiwork less handsome, or the sound of his CD player less good than that of any other unit reviewed here. It's just that the realities of the industry are not on his side.

Harman/Kardon HD7600

Harman/Kardon Incorporated, a Harman International Company, 240 Crossways Park West, Woodbury, Long Island, NY 11797. HD7600 compact disc player with remote control, $599.00. Tested sample on loan from manufacturer.

I should have reviewed this excellent CD player in the last issue—that's how old it is, with an updated Mark II version already in the pipeline—but then I decided to consolidate all CD playback-related material in this issue. (The same design also exists in a somewhat leaner $449.00 economy version, called the HD7500, which is identical except that it lacks digital outputs, index search buttons, and a few other minor control conveniences—almost forgivable at a list price $80 lower than that of the Carver above.)

This "old" HD7600 represented my first encounter with any of the new 1-bit DAC architectures—in this case the Japanese MASH system—and I was duly impressed with the low-level linearity. Since then I have looked at other "bitstream" implementations, and in every one of them the nonlinearity was within the accuracy limits of my test setup. Better than trim pots, right? As for the Harman/Kardon analog stages, they are the most elaborate to be seen in any CD player made in Japan, regardless of price, with fully discrete circuitry and total observance of the guidelines laid down by Matti Otala when he was with the company years ago. Unfortunately, that approach is inherently more costly than the conventional one using inexpensive chips, necessitating inevitable compromises in the quality of parts and construction to keep the price of the HD7600 down. One wonders whether high-performance integrated op amps wouldn't have been a more appropriate choice in this price range. Of course, some of that Harman/Kardon personality would have been gone. In any case, I measured no anomalies worth mentioning in either the digital or the analog circuits.

Ergonomically I found the HD7600 to be excellent; the buttons are well-located and have a nice, positive feel, as does the disc drawer; the display is first-rate. The sound? Again, no different from that of any other unit in this group, as far as any of us could tell. At $599, however, this is an outstanding buy, and the $150 cheaper HD7500 version is a very serious bargain if you can live with its austerities. The Mark II versions have been announced at $699 and $529, respectively; at those price points they are somewhat less attractive and begin to have more competition, unless the improvements are greater than I'm currently aware of.

JVC XL-Z1010TN

JVC Company of America, division of US JVC Corp., 41 Slater Drive, Elmwood Park, NJ 07407. XL-Z1010TN compact disc player with remote control, $700.00. Tested sample on loan from manufacturer.

JVC is also beginning to make the 1-bit DAC scene, so I don't predict a long life for this somewhat aging multibit unit. What they call its "quadruple full-time linear 18-bit combination D/A converter" is actually the lowest-grade Burr-Brown 16-bit DAC, the PCM56P, with external components added to handle the lowest 2 bits, bringing the count to 18. Something of a kluge. The low-level gain linearity I measured was essentially perfect in one channel and about 1/2 LSB off in the other channel (at –90 dB), so I can't complain too much. A sample-and-hold circuit is needed with this DAC configuration, and there are other little not-quite-audiophilic touches throughout, such as an electrolytic capacitor in series with the output signal and the plebeian NE5532 op amps in the analog stage. I also saw a bit more RF in the output than I liked. On the other hand, the so-called K2 Interface, designed to remove jitter by reclocking the digital data before D/A conversion, is a state-of-the art feature, made less impressive only by the fact that Sony does the same thing without making a fuss about it.

Don't misconstrue these critical observations as a general disrecommendation of the JVC. If no other CD player were available to me, I would find it eminently satisfactory. It has everything; it does everything; and it sounds as good as any of the others reviewed here. But in a circuit-for-circuit, chip-for-chip, feature-for-feature shoot-out with the competition in its price range—not neglecting mechanical construction and controls/ergonomics—it is clearly outgunned. JVC has considerable sophistication in this area of electronics, and I fully expect them to come out with something more exciting in the very near future. They have already announced a "pulse edge modulation" 1-bit DAC that looks promising in their lower-priced line.

Meridian 208

Meridian America Inc., 14120-K Sullyfield Circle, Chantilly, VA 22021. Meridian 208 compact disc player and preamplifier, with 209 remote control, $2950.00. Tested sample on loan from USA distributor.

Yes, I know, the English gave us Shakespeare, Newton, the RAF fighter pilots of 1940, and the Beatles. But they drive on the left side of the road, serve lukewarm beer and hang their plumbing on the outside of the house so the water freezes in the winter. They're too damn impractical, and so is their Meridian 208. I still can't believe it costs almost $3000, even with the built-in rudimentary line-level preamp and Boothroyd Stuart pin-striped styling. (For pin-stripe in four figures, I'll take Giorgio Armani.)

Seriously, though, I had an off-putting experience as soon as I tried to put the battery into the remote control unit. Everybody else's battery compartment has a sliding or snap-on cover, but not Meridian's, chappies. Four Phillips-head screws keep it in place, torqued down by some Kentish lout with large hands, and the screw heads are butter soft so that the resistance to any counterclockwise turning will strip them. Unless you enjoy the sight of metal shavings shedding from your new $2950 toy, your heart will be filled with hatred by the time you have that silly, impractical cover off the battery compartment. A little intelligent planning could have saved a lot of aggravation here. And that's not all. The display of the 208 is of the earliest, minimalist Philips design, the same that Tandberg used to have, and the control buttons are an ergonomic fiasco. For example, moving to an index point is a three-step operation. Of course, inconvenience is the proof of high-end performance in some circles.

Well, what about that performance? Quite impeccable, I must admit. The Philips Bitstream 1-bit DAC architecture used results in just about perfect low-level linearity, probably more perfect than my measurement setup. Although the Philips SAA7321 chip is a stereo DAC, the 208 has two of them connected in a balanced configuration, a good idea made somewhat less remarkable by the fact that the Harman/Kardon HD7500 economy CD player at less than one-sixth the price uses a similar arrangement. What's more, the latter gives you discrete analog circuitry whereas the 208 uses the NE5534 integrated op amp, the current Philips standard. On the other hand, the output of the 208 doesn't pass through an electrolytic capacitor like that of the HD7500 but is directly coupled with a DC servo. (Be thankful for small favors.) Anyway, regardless of the strange mixture of chintzy and quality touches on the circuit board, I could measure no imperfections worth mentioning in the electrical performance of the 208. As for mechanical performance, the disc transport is basically the same as in the top-of-the-line Philips LHH1000—no complaint there—but the latter is a far more deluxe and better engineered package.

That leaves us with the $2950 question: What does the Meridian 208 have going for it at its price? The answer is—Martin Colloms! In the January 1990 issue of England's *Hi-Fi News & Record Review*, a few months before the 208's American debut, Mr. Colloms canonized it as the overall best-sounding CD player known to him. What can I say? It certainly sounds as good as any of the others reviewed here, but better? Not to my or my associates' ears. Of course, Martin didn't use the ABX Double-Blind Comparator, and I did. So the best I can say for the 208 is that I'll gladly use it in my system as long as (1) somebody else pays for it and (2) somebody else operates the buttons.

Onkyo Integra DX-7500

Onkyo U.S.A. Corporation, 200 Williams Drive, Ramsey, NJ 07446. Integra DX-7500 compact disc player with remote control, $700.00. Tested sample on loan from manufacturer.

This is a very impressive player for the money, with a surprising number of quality features, although one of my routine tests tripped it up in a mysterious way. More about that in a moment. What I like about the DX-7500 is that it is very solidly built and generously engineered, with a high-quality disc transport employing linear tracking (but, inexplicably, without a velocity-sensing feedback coil), control buttons with a positive feel and excellent ergonomic layout, dual transformers and optocoupling to isolate the digital and analog blocks, coaxial and optical digital outputs, DC-coupled analog output with servo, and a choice of several line-level output options, one of which (labeled "direct") allows the signal to pass through only a single analog signal-processing stage. Quite a serious piece of equipment.

The DAC is the J grade of the 18-bit Burr-Brown PCM58P, with all four of the optional linearity adjustments included. The trim pots appeared to be more or less correctly adjusted in my sample, with low-level errors ranging from 1/4 to 1/2 LSB, including my measurement errors. I have seen slightly better J-grade performance, but only without the trim pots! An interesting though probably not very significant measurement showed the usual clipping of a 0 dB square wave by the digital filter to be symmetrical through the "direct" output but asymmetrical through the "fixed" output. Decisions, decisions.

The one thing I didn't like was the response to the high-frequency two-tone test (19 kHz at –6 dB plus 20 kHz at –6 dB, equivalent to a single tone at 0dB). All players passed this test without any intermodulation products within the audio band, but the Onkyo showed intermittent distortion blips of the order of –40 dB (1 %) at constantly shifting frequencies (3.6 kHz was typical) and of short duration, impossible to pin down with any degree of accuracy. I have no idea what this was, certainly not garden-variety steady state IM. I was ready to make the statement that nobody needs a better CD player than the Onkyo Integra DX-7500, but this anomaly stopped me. That doesn't mean that any other player beat it in the ABX comparisons; it held its own against all comers, but so did all comers against it.

Philips CD-80

Philips Consumer Electronics Company, One Philips Drive, P.O. Box 14810, Knoxville, TN 37914-1810. CD-80 compact disc player with remote control, $799.95. Tested sample on loan from manufacturer.

From where I'm sitting, Philips appears to be headed for across-the-board 1-bit DAC architecture in their CD players, even though they originally developed the concept with portable and low-end models in minds. Thus the 16-bit CD-80, representing the highest evolution of the line in which the CD880 used to be the corresponding model, may very well be the last of the Mohicans; there seem to be no newer multibit models coming out of Hasselt, Belgium. In any case, I find the CD-80 to be an outstanding piece of equipment, with virtually no negative attributes.

The unusually heavy chassis is very similar in construction to that of the $4000 top-of-the-line LHH1000, and many of the electronic components are of unexpectedly high quality. The disc transport is not quite the same as in the LHH1000 but is still excellent. Special attention has been paid to routing the digital and analog signals away from each other. Fully discrete double-regulated power supplies are another feature. The front-panel control layout is somewhat "creative" and takes a little getting used to, but you soon learn to like it. The remote control is conventional.

The chip complement includes, among others, the TDA1541A S1 "Golden Crown" DAC (which is designed without external trimming options), the SAA7220P/B digital filter, and the NE5534 decompensated op amp, a combination also used in the LHH1000. The analog outputs are direct-coupled with a DC servo and protected against failure of the servo. I found the worst-case low-level gain-linearity error (in the less good channel at –90 dB) to be of the order of 1/2 LSB, which is not quite as good as can be obtained with the best Burr-Browns but certainly good enough. The same can be said of the low-level harmonic distortion spectra; there are some tiny blips peeking out of the noise floor where there are none with the best Burr-Browns, but again nothing worth complaining about. Various high-frequency test signals produce the usual predictable small-amplitude beat tones above the audio range; that's the way Philips chooses to do the filtering and there's nothing really wrong with that, either. I would have liked, however, to see less RF coming out of the back end of my sample, although it didn't create any problems in my electronic environment.

The bottom line is that the CD-80 has made all those audiophile Philips mods overpriced and irrelevant. Philips took a good look and included in their package pretty much all of the goodies the modifiers had come up with, only with slicker execution and at a better price. I could live with this player, even if the future doesn't belong to 16-bit DACs.

Pioneer Elite PD-71

Pioneer Electronics (USA) Inc., P.O. Box 1720, Long Beach, CA 90801. Elite PD-71 compact disc player with remote control, $850.00. Tested sample on loan from manufacturer.

This is another CD player I'm catching at the tail end of its natural life; by the time you read this it may have been phased out and replaced by the almost identical PD-73. The only difference is that the latter has the new 20-bit Burr-Brown DAC, on which Pioneer has a temporary exclusive (at least as far as I know). The PD-71 uses the Burr-Brown PCM58P-K, which is the top grade of the top 18-bit DAC in the line. You can assume that the PD-73 will perform similarly, or possibly even better if the jump from 18 to 20 bits is indeed meaningful, but only if the MSB trim pot in each channel is accurately adjusted for maximum DAC linearity. Of the two samples of the PD-71 I had a chance to look at, one appeared to be trimmed in perfectly, the other not. Unfortunately, it was the latter that became my long-term lab sample; the first I had checked out on the fly while its owner waited. It would be meaningless to specify the errors in the poorly adjusted sample, as it would provide no information about any other sample, so let me just say this:

The Burr-Brown PCM58P-K, with the optional external bit-linearity trimming correctly fine-tuned, is the most nearly perfect 18-bit DAC on God's green earth, so the low-level performance of your PD-71 (or, by extension, the upgraded 20-bit PD-73) has nothing to do with engineering and is strictly dependent on the individual history of your unit, starting at the QC stage. That doesn't really help you, I know. It's fairly easy to make those adjustments on a properly equipped lab bench, but I'd be surprised to find a dealer able and willing to do it. The answer is, of course, to design the equipment without the trim pots.

The PD-71 has many other quality features in addition to the potentially best multibit DAC there is; in fact, it's more or less the Japanese opposite number of the Philips CD-80, each offering minor advantages over the other. The Pioneer is more deluxe in appearance, with one of the most elegant cabinets in the business; the Philips is much heavier, more rugged-looking, more technocratic/industrial in style. Ergonomically the Pioneer is excellent and perhaps easier to like; I happen to prefer the Sony CDP-608ESD to either one. The Pioneer has an excellent linear-tracking disc transport made with Sony components; there is some tenuous evidence that this transport tracks certain defective discs that the Philips has trouble with. The Pioneer appears to have slightly better RF suppression; it uses discrete push-pull power supply regulators (good); it operates all active op amp stages in the inverting mode to eliminate common-mode input signal distortion (good); unlike the Philips, it has no provision for headphones nor a volume-controlled output (high-end etiquette); it passes the analog signal through three gain stages and a nonpolar electrolytic output capacitor (not so good—the Philips does it better); it also uses some op amps that aren't exactly state-of-the-art. Overall, the PD-71 is very conscientiously built and worth its price.

As I indicated in the Carver review above, the PD-71 didn't sound verifiably better or worse than the much lower-priced Carver TL-3220, or any other CD player reviewed here, but that's an old refrain of this survey by now. Even so, I'd rather own the Pioneer as long as I get a chance to adjust those trim pots.

One more thing. The PD-71 is Precision Audio's current favorite for their excellent D1 Analog mod (see Issue No. 12). The debut of the PD-73 presages heavy discounts on leftover PD-71's if you're interested in taking that route.

PS Audio "Digital Link"

PS Audio, 302 South 13th Street, Grover City, CA 93433. "Digital Link" outboard D/A converter, $799.00. Tested sample on loan from manufacturer.

Paul McGowan and Bob Odell of PS Audio are decidedly in the tweako camp. They believe in a lot of the audiophile voodoo I ridicule in these pages (the stuff that I can't hear and nobody can prove is hearable). At the same time, their products are superbly engineered and utterly sensible in terms of cost-effectiveness. As long as they make equipment like the Digital Link, they can go ahead and believe in the tooth fairy for all I care.

As a matter of fact, if an enlightened audio perfectionist made a wish list of D/A processor design elements and features requiring no extravagant expenditure, he would be describing something reasonably close to the Digital Link. It's basically what the demanding but sane aficionado needs if he decides to take the outboard processor route.

The DAC around which the processor is built is the Burr-Brown PCM61P, which according to some maverick practitioners (but not Burr-Brown) is more linear than the PCM58P and "sounds better." The J grade of the PCM61P was used in my sample; later production, according Paul McGowan, uses the K grade. Regardless of these distinctions, I measured essentially perfect low-level linearity in both channels of the Digital Link. The analog section consists of a single stage employing a state-of-the-art, high-speed, complementary bipolar IC (Analog Devices AD847) in a rather unusual low-feedback configuration. A high-current output buffer circuit is used in conjunction with the IC, and the output is direct-coupled with only an offset pot for DC cancellation. (No copycat, this Paul McGowan.) The current-to-voltage conversion circuit is passive, resulting in the relaxation of settling-time requirements for the op amp (good), greater voltage swing on the DAC output line (dangerous living distortionwise), and a significantly decreased signal-to-noise ratio (not so good, but more about that in a moment). The power transformer is a bit on the chintzy side but is separated from the tiny main chassis by an "umbilical cord" in order to reduce hum levels; the Digital Link and the Aragon D2A were the only units in these tests with that feature. No jitter attenuation circuit is used in the SPDIF decoder, an economy I deem acceptable considering where the money was spent instead. On the other hand, I wish there were an optical input in addition to the coaxial one provided; fear of optical data transmission is a high-end neurosis I can't relate to. Physical construction is unimpressive; the crowded PC board floats inside the generic modem chassis and is kept from shorting to the chassis on the foil side by a thin piece of plastic.

My laboratory tests revealed no objectionable weakness anywhere. Yes, the noise floor in the absence of a digitized signal is a whole order of magnitude (approximately 20 dB) higher than in the best Japanese CD players, but in the presence of a signal it is just about the same, rendering the issue academic. The maximum signal-to-noise ratio achievable on a CD with 16-bit A/D encoding is theoretically 98.1 dB, and the Digital Link accommodates that with a very healthy margin to spare. I must therefore approve of the audio-quality-oriented noise-floor trade-off. I was less happy about the amount of RF I saw pouring out of the back end of the unit, but I can't report any interference problems in my particular electronic environment (for whatever such a limited criterion is worth).

Bottom line: I left the Digital Link in my reference setup, driven by whatever transport happened to be handy. No, not because it sounded better than any other unit reviewed here—as you already know, it didn't, and vice versa—but because it *ought to* have sounded better on the basis of its analog engineering and intelligent trade-offs. Let no one say I've stopped being a hi-fi nut.

Sony CDP-608ESD

Sony Corporation of America, Sony Drive, Park Ridge, NJ 07656. CDP-608ESD compact disc player with remote control, $900.00. Tested sample on loan from manufacturer.

When I received this outstanding unit for review, I made the mistake of saving it for this survey instead of squeezing it into Issue No. 14. Between Sony's propensity for constant model changes and this journal's propensity for delayed publication, that turned out to be a bad decision. The 18-bit CDP-608ESD has now been replaced in the Sony line by the 1-bit CDP-X55ES, at the same price; indeed, their entire premium ES line now uses 1-bit DACs, of a newer generation than either Bitstream or MASH. Even so, some dealers probably still have a CDP-608ESD or two left in their stockroom, and if the closeout discount is deep enough, grab it—it would be a very good value.

Frankly, I don't see how the new Sony 1-bit DAC can improve on the performance of the Burr-Brown PCM58P-J used in this superseded model—without MSB adjustment, I might add. I measured almost unbelievably perfect gain linearity all the way down to the lowest levels and saw no harmonic distortion blips peeking out of the noise floor at any level. The other star performer of the act is the Sony CXD1244 digital filter chip; here the design is one up on others by combining 8 times interpolation with noise shaping and digital de-emphasis. (The advantage of digital de-emphasis is that it removes several passive components and a FET switch from the op amp summing junction; the disadvantage in comparison with analog de-emphasis is possibly increased noise and distortion at the player's output when playing a disc with emphasis.) Another advanced feature is that the digital data is reclocked before entering the DAC. (JVC built its entire "K2 Interface" promotion around the same jitter-correcting technology.) The disc transport assembly is Sony's high-quality G chassis, which combines a lightweight electromagnetic linear drive with the KSS151 three-beam optical block on a marble-like resin base. The decoder chip is the CXD1165, which was the *dernier cri* when the player was introduced. It's all pretty high-tech for a model going out of style and makes you wonder whether the switch to 1-bit DAC architecture is an engineering or a marketing decision.

The analog section, on the other hand, is not nearly as perfectionistic. The so-so NE5532 op amps are used as the active elements; the supply rails are ±12 volts instead of the ±15 volts specified by Signetics; the output signal passes through an electrolytic capacitor (albeit bypassed with a small film capacitor)—in other words, Japanese upper-mid-fi audio circuitry. Not that it made an audible difference in double-blind comparisons at matched levels against various other players in this group; my objections are therefore somewhat theoretical.

Having said my piece on that, I must commend the CDP-608ESD for two things: the front-panel control layout and display, which were my ergonomic favorites among all the units reviewed here, and the total absence of RF at the output. Sony is the king of digital audio—if only they lavished equal care and talent on the analog part!

Sony D-555 "Discman"

Sony Corporation of America, Sony Drive, Park Ridge, NJ 07656. D-555 "Discman" portable compact disc player, $450.00. Tested sample on loan from owner.

Does this yuppie toy, conceived for headphone listening on the way to the brokerage office or the sushi bar, belong in the company of the audiophile-caliber CD players reviewed here? I definitely think so. Even though its measured performance turned out to be somewhat inferior to that of any other unit in this survey, a quick ABX shoot-out between it and the $2950 Meridian 208 failed to reveal any reliably identifiable differences in sound. Admittedly, this test didn't go to the usual 16 successive double-blind trial runs but stopped about halfway. But even a sloppy nonblind comparison of the Discman with the Onkyo Integra DX-7500 was quite inconclusive. The Sony sounded maybe a little grittier and the Onkyo smoother to the listeners, who were quite happy that they didn't have to do any blind identifying. The visiting owner had to take the player back home with him, so I can't swear that a month's worth of comparisons wouldn't have established some clearer differences. At any rate, no other CD player, regardless of price, "blows away" the D-555 in sound quality. That much is certain.

The highly touted Digital Signal Processing (DSP) features of the Discman I found less impressive. The bass boost below 80 Hz (useful on cheap headphones), dynamic range compression (for car listening), "surround sound," and graphic equalization at five frequency positions—all of it digitally implemented—seem quite rudimentary; for example, the so-called surround-sound processor appears to equate adjustable phasiness with an actual surround effect. Of course, shoehorning even such rudimentary processing capability into that itty-bitty box is an achievement in itself, and that neat little LCD readout is a nice touch.

On the lab bench a number of little weaknesses became apparent. The noise floor is only 90 dB below the 0 dB reference level, whether or not a digitized signal is present. The de-emphasis is not very accurate; it's off by as much as 1 dB at some frequencies and isn't right on the button at any frequency. The low-level linearity is simply not in the same league with the standard-sized players; in the less good channel I measured a full 1 LSB error at –80 dB, and at –90 dB there was total chaos in both channels. The resolution of the Discman is in effect somewhere between 14 and 15 bits—not a full 15 bits for sure—although the DAC is specced at 16 bits and paired with an 8 times interpolating digital filter. The digital circuitry even provides an optical output, but no coaxial. It should also be noted that the D-555 inverts the polarity of the input signal, a rare quirk these days. As for the disc drive and optical block, the quality seems good enough, but I was unable to play track 56 of the 99-track Philips "Audio Signals Disc 1" (SBC429); the damn thing kept muting for no discernible reason. Another possible cause for concern is the question of reliability and servicing; the owner of the D-555 I tested had had a terrible time with its predecessor, the D-T10; in fact, the D-555 was Sony's please-don't-bother-us-anymore free replacement for the hard-to-repair D-T10. Hm.

All in all, however, I must come out in favor of the Sony Discman. It's small, it's cute, it's not overpriced for what it offers, and it works. Its measurable shortcomings are almost certainly below the audible threshold, and its impact as audiophile costume jewelry is undeniable.

Theta DS Pre Basic

Theta Digital Corporation, 5330 Derry Avenue, Suite R, Agoura Hills, CA 91301. DS Pre Basic digital signal-processing preamplifier, $2400.00. Tested sample on loan from manufacturer.

Last in this survey, because of a vagary of the Roman alphabet, is the unit with the most elaborate digital circuitry. The DS Pre Basic isn't actually a full-fledged preamplifier; it differs from a monolithic D/A processor only in that it has a digital tape loop, inputs for a single line-level analog source, fixed analog tape outputs in addition to the variable analog line outputs, a volume control, and a balance control. It could be the control center for a simple CD/tuner/tape system, but that's it. The volume/balance controls involve no additional active stage. And yes, there's no optical input; it would spoil the high-end image, don't you see? (I don't.)

Mike Moffat, the designer of Theta equipment, is yet another of those strange technologists who *do* everything knowledgeably and scientifically but *talk* voodoo to wide-eyed audiophiles. (Do they think it's good for business, or do they with great sincerity manage to work themselves into a schizzy Jekyll and Hyde mind-set?) Mike Moffat's software-driven digital filter for the DS series, in effect a single-purpose computer, is possibly the best-designed and most advanced in the business, at least as judged from his conversations with David Rich (no schematics and no computer code being available). At the same time, Mike claims to hear differences between printed circuit boards made of different materials and between DAC chips whose pins are made of different metals. Are these claims based on double-blind comparisons at matched levels, Mike? Huh? (On the other hand, nobody held a gun to my head to make me go into high-end audio and take all this guff.)

Other than the digital filter, which is implemented on a general-purpose DSP chip, the chip complement of the DS Pre Basic isn't proprietary but a little different nonetheless. The 18-bit DAC is the AD1860N-K from Analog Devices, the K suffix indicating the highest grade. A trim pot in each channel adjusts the MSB. The op amp for the analog signal is the PMI OP42, a high-quality but no longer state-of-the-art IC (a generation behind the AD847 that PS Audio uses, for example). The output is direct-coupled with a DC servo. Construction and parts quality is very high (except perhaps for the use of that op amp), making the price tag palatable to the critical purchaser.

One design parameter that obviously matters a great deal to Mike is jitter. The SPDIF decoder includes three phase-locked loops to reduce jitter. The PLL circuits used are inexpensive CMOS devices (CD4046) rather than the more expensive crystal-controlled PLL circuits used in many Japanese decoders, which theoretically yield even lower peak jitter. Even so, Theta is the only company other than Mondial/Aragon to publish a peak jitter spec for the SPDIF decoder, namely 1 ns. Nice.

My lab-bench and in-use experience with the DS Pre Basic had its ups and downs. In the first sample they sent me, the left and right outputs were reversed and the MSB trim pots misadjusted. I proved to myself, however, that the latter could be adjusted for virtually perfect low-level linearity. A second sample, which looked more like a finalized production unit, had the left/right error straightened out, but the MSB adjustments were still way off, and the low-level linearity was poor until I again tried and succeeded tickling it into perfection. A few weeks later I sent the second unit back to Theta for a "basic capacitor upgrade" they had just then put into production. At the same time I insisted that they run the unit through QC again to bring it up to their highest standard MSB-wise. When it came back, everything looked textbook-perfect on the lab bench, with no anomalies or deviations worth mentioning. All right, one peculiarity, not a complaint: square waves at the 0 dB level are clipped more radically by the digital filter than in any other D/A processor I've looked at. No big deal. Also—and this has nothing to do with D/A conversion accuracy or audio fidelity—there's a lot of RF in the output, although a little less than before that latest mod. Still, it interfered to a slight degree with the performance of an FM tuner about three feet from the Theta. Little gurgles, hash, and birdies.

As for the sound of the DS Pre Basic—great! The fact that it's indistinguishable in ABX tests from the sound of any of the other units reviewed here doesn't make it less great. Indeed, on the basis of general engineering sophistication and palpable quality on the circuit board, this is arguably the unit of choice in this survey, but those pesky MSB trim pots and the RFI give pause. My impression is that Mike Moffat has the ability to engineer a flawless piece of equipment; however, Theta as a manufacturing/marketing operation needs to get its act together a little more reassuringly before I can be entirely satisfied. For openers, they could include an instruction booklet.

Recommendations

As the Germans say it rhymingly, *die Wahl ist eine Qual.* The choice is agony. Trade-offs, trade-offs, nothing but trade-offs. My ideal CD player, or transport/processor, would have the nice feel, ergonomics, and quality control of the Sony CDP-608ESD... the digital filter of the Theta DS series... the 20-bit Burr-Brown DAC of the new Pioneers (in the K grade but without MSB adjustments)... the construction-quality-to-price ratio of the Aragon D2A and maybe even its discrete analog circuitry (but with the addition of an output buffer)... I could go on but what's the use? It will never happen. Read David Rich's article, read my reviews, and decide what suits you best. When the unequivocally best choice arrives, I'll be less equivocal. Meanwhile (chuckle, chuckle) "they all sound the same"—and if you disagree with that, don't just vituperate but prove the contrary in a double-blind comparison at matched levels. ◊

The Wire and Cable Scene: Facts, Fictions, and Frauds Part I

By Peter Aczel
Editor and Publisher

This is just a curtain raiser to get you in the mood: an almost two-year old background story, told here for the first time "like it is." The technical/critical examination of the subject is coming next.

The protagonist of this story is David L. Clark, noted Detroit audio consultant under the name of DLC Design, guiding spirit of the Southeastern Michigan Woofer and Tweeter Marching Society, and notorious designer (with others) of the diabolical ABX Double-Blind Comparator. He is of course also known to our readers as that wry but rational voice on our Seminar 1989 panel. Dave should really be the one to be reporting here on his own work, but he is letting me do the talking because he is, as he says, a total burnout on the subject as a result of the apathy, insincerity, misrepresentations, and downright hostility he had encountered.

I must point out up front that Dave's story is not only about wires and cables but also (perhaps even primarily) about power amplifiers, but the latter are not our concern here for the moment. What happened was that Dave had approached the Audio Engineering Society to suggest using their 85th Convention in Los Angeles in November 1988, with its many knowledgeable and highly motivated participants, as an opportunity to run some serious double-blind listening tests in a workshop to be called "Esoteric Audio—Can You Hear It?" The way I understand it, the AES consented enthusiastically and promised three listening rooms, which they delivered, as well as nine assistants (three per room, to work in shifts), which they did not—not even a single one. Strike one against Dave; he was left to run the show more or less by himself, sink or swim.

Luckily, he didn't give up, enlisted a number of interested helpers, and managed to complete 434 speaker-cable comparison trials and 659 power-amp comparison trials over a two-day period. (Each listener opted for anywhere between 2 and 7 trials in cables and either 2 or 3 trials in amplifiers.) On the climactic third day there was a panel discussion of the results, of which a cassette recording is available, documenting voices as thoughtful and informed as those of Richard Greiner, Floyd Toole, and John Vanderkooy, and as tweaky and undisciplined as that of Michael Fremer. What emerged quite clearly from the discussion was that the flat-earthers and cargo cultists were not about to allow the facts to overturn their belief system. Strike two against Dave, and another reason for his frustration.

All I want to do here is to present the givens and the results of the speaker cable tests factually (this has not been done anywhere so far) and to comment very briefly on the meaning of the outcome.

Cheap industrial wire vs. Monster Cable's best.

The high-end speaker cable chosen for the double-blind listening comparisons was Monster Cable M1, which at the time was that highly promoted company's top-of-the-line model, at nine dollars per foot. (Since then they have come out with the "Sigma" insanity at almost five times that price.) According to Dave Clark, the Monster Cable people were then asked what they considered to be an absolutely unlistenable cheap cable of roughly the same gauge. Their answer was 10-gauge THHN industrial wire, so that was chosen as the low-priced foil for the M1. To give each cable the chance to assert its sonic personality, if any, 30-foot lengths were compared, switched by hand (no ABX box!) between a Perreaux amplifier and a pair of Tannoy "Dual Concentric Studio Reference Monitors."

Now it should be noted that 434 trials by 86 persons (the very few persons who came back to try again are counted as two) constitute a respectable statistical base permitting reasonably accurate conclusions. The score: 207 correct identifications out of 434 same/different trials; in other words, the participants were "batting" .477 in their attempt to identify the cable they were listening to as being the same as, or different than, the one before. Sheer guessing would be expected statistically to result in .500, virtually the same figure. So, to put it bluntly, *no difference in sound* was heard between the audiophile Monster Cable and the 10-gauge cheapo wire, not even with 30-foot lengths.

The reaction to this outcome by the high-end religionists was of course predictable. Sure, they said, the unwashed masses can't hear the difference, but we golden ears can—look, here's somebody who got 6 right out of 6, here's 6 out of 7, here's a 4 out of 4, and so forth. All right, class, what's wrong with that argument? That's right, we're dealing with the bell-shaped (Gaussian, or normal) distribution curve. In any large sample of a totally random process, a very few individual readings will be all the way to the left under the flare of the bell (lowest scores), and a very few will be all the way to the right under the flare (highest scores). For example, if you toss a hatful of coins 1000 times, you could get all heads once or all tails once, but most of your tosses will come out half heads and half tails, or not far from that. There were several 1-right-out-of-7 scores at the AES, in Gaussian balance with the high scorers. In one or two instances, the *same* person had a high score in the morning and a low score in the afternoon. Thus 86 deaf-mutes guessing wildly about the cables might also have come up with a 6-right-out-of-6 score sheet among them. No, amigos, a convincing performance by a golden ear would have been, say, 4/4 in the morning, followed by 6/6 in the afternoon, followed by 7/7 the next morning. Either there is a difference in sound, in which case a golden ear will reliably hear it, or there isn't. Let's face it, there wasn't.

I can't blame Dave Clark for feeling disgusted after seeing his quite unexceptionable methods and conclusions met with denial by the emotionally vested interests of high-end audio. Fortunately there was no "grungy" ABX box on which to blame the obscuration of the alleged sonic differences. That would have been too pat. The denials had to take a more convoluted, whiny, philosophically petulant course—witness John Atkinson's remarks from the floor side of the panel discussion and his report on the workshop in the "Industry Update" column of the January 1989 *Stereophile.* What I don't understand is why these subjectivist diehards never ask the obvious question of whether there exists any kind of mechanism whereby A and B *could* differ in sound. Were the two cables sufficiently different in resistance and/or capacitance and/or inductance to interact quite differently with the source impedance and terminating impedance, and thereby generate significantly different transfer functions? But no—that's not what they ask. They look at the brand names, they look at the prices, and they just know which one sounds better. Part II of this series will address in depth the electrical network characteristics of speaker cables and the resulting transfer functions at the amplifier/speaker interface. ◊

Hip Boots

Wading through the Mire of Misinformation in the Audio Press

***Editor's Note:** This column was absent from Issue No. 14, and in Issue No. 13 it was still using the editorial "we." Even though signed articles and the first person singular have now been introduced as the standard convention of this publication, I feel no need to sign my name here, since no one is likely to attribute this sort of thing to another author. Rest assured, the curmudgeonly "I" is I.*

Willie Sutton robbed banks "because that's where the money is," as he explained, and I keep coming back in this column to *Stereophile* because that's where the audio misinformation is. That's not the only place, I must admit, but then it's not the only publication I take to task here, either.

John Atkinson in *Stereophile*

In the "As We See It" leadoff editorial column of the May 1990 issue of *Stereophile*, John Atkinson defends at great length the Santa Fe magazine's practice of measuring loudspeakers at the altitude of 7000 feet above sea level, in response to the doubts I expressed about the validity of such measurements in Issue No. 14 of **The Audio Critic**. He writes that I feel the need to defend my reputation by attacking *Stereophile*'s, and after several pages of circuitous arguments and frequency response curves (the thrust of which is that, yes, altitude makes a *small* difference, but so what) he concludes that "Mr. Aczel's hypothesis"—which he misstates so outrageously that I refuse to quote him for fear of giving the misstatement permanence—"is incorrect."

Now, in my Madison Avenue days, I used to have a sign in my office that said, "I may have my faults but being wrong isn't one of them." So you can imagine how much it goes against my grain to refrain from punching holes in JA's technical arguments (and how disappointed he will be), but a larger issue than that needs to be addressed here, namely intellectual honesty in audio journalism. You see, JA is hypocritically responding only to the *lesser* of the two examples of altitude-skewed speaker testing I cited (viz. Waveform) and carefully avoids any reference to the big, embarrassing one (viz. Carver). It's the same as if, for example, Nixon were making eloquent excuses for the Agnew scandal and pretending that Watergate never happened.

I'll come back briefly to the Waveform supertweeter's alleged peak. First, however, I must remind JA that the unmentionable Carver "Amazing Loudspeaker," in the early Platinum Edition reviewed by Dick Olsher in the February 1990 *Stereophile* (as distinct from later versions), was in effect two totally different speakers at sea level and at 7000 feet—or so I am told by its designer, who was very unhappy about the difference and began to take steps to fix it as soon as he became aware of it. The review glossed over this problem and treated the consequent deficiencies of the speaker as engineering ineptitudes, amusing eccentricities, and poor quality control. According to Bob Carver, he begged DO and JA to audition and measure the speaker at a less extreme altitude, but they flatly refused. That puts a very different complexion on the subject than JA's hairsplitting little apologia. I'm now firmly convinced that before the Carver experience no one at the magazine had the slightest awareness of the altitude problem and afterwards the problem had to be declared insignificant to prevent all sorts of skeletons from tumbling out of the closet. Somehow, "the truth, the whole truth, and nothing but the truth" is considered bad for business in Larry Archibald territory, as I have pointed out before.

To return to the Waveform speaker system, here are the plain facts: The National Research Council laboratory in Canada measured one sample of the system to be very flat. I measured another sample to be very flat. In Santa Fe, at 7000 feet, they measured the system (I don't know which sample—not mine) to have a big 16 kHz peak. Then JA had an old, spare sample (not taken from any of these systems!) of the suspected Philips ribbon-type supertweeter measured in Los Angeles at sea level and in Santa Fe at 7000 feet. In the tiny, hard-to-read graphs accompanying his May editorial, I discern approximately 2 dB more output at 16 kHz in Santa Fe than in Los Angeles, exactly as one would expect. He calls it 0.4 dB. (Sure, John, if you say so...) Anyway, how does this cockamamy experiment put a Q.E.D. on the general unimportance of the altitude issue, and how does it prove me "incorrect"? The mind reels.

One more thing. In a nasty footnote to the July 1990 "Industry Update" in his magazine, John Atkinson has this to say about my participation in the 8th International Conference of the Audio Engineering Society, which took place in May in Washington, DC: "Characteristically, Mr. Aczel avoided discussion of his role [as audio reviewer], choosing instead to attack the other high-end magazines." He is referring to a special session on the reviewing of audio products, where I was alphabetized to be the first panelist to make an introductory statement.

Again, being "characteristically" ill at ease with the whole truth, JA omits that all of us panelists, including me, discussed our roles as audio reviewers all evening, only my *opening* remarks were not about myself but about the general subject of *accountability* in equipment reviewing—and that made JA squirm because it set the theme for the next hour or so. He ran into a bit of trouble trying to explain to a large roomful of the top academics and professionals in audio why he and his staff can't prove what they claim to hear. I suppose it did seem like an attack to him, just as my comments on speaker measurements at 7000 feet above sea level seemed like an attack on his magazine's "reputation." Anyone sitting there in that Washington conference room could see, however, that some of the best brains in audio had serious doubts about that reputation to begin with.

Neil Levenson in *Fanfare*

Fanfare calls itself "The Magazine for Serious Record Collectors," and in my opinion it lives up to that tag line. Published six times a year in the form of a fat paperback book, sometimes exceeding 500 pages, it is unquestionably a quality publication, unapologetically highbrow and strictly of a "scholarly" typographic format (i.e., every page looks the same). If a labeled classical recording exists at all, you can be fairly certain that it will be reviewed in *Fanfare*, not only when first released but also whenever repackaged in some other form or combination. Furthermore, the review is more likely to be intelligent and musically enlightened than those in the newsstand magazines, since Editor/Publisher Joel Flegler has a large number of mostly excellent freelance specialists at his beck and call and lets them write as they please. They're a pretty sophisticated bunch. The more's the pity, then, that Mr. Flegler chose as his Audio Editor a self-indulgent pseudo expert like Neil Levenson.

Not that I expect Mr. Flegler to be an expert on audio experts. He is obviously a music man (and a good one), perhaps without any close acquaintances in the inner circles of audio. But it's possible for a resourceful nonexpert to push some buttons, beat some bushes, make some waves, and find a genuine expert with genuine credentials. Mr. Flegler got snookered, I think, by a fellow music devotee—because Neil Levenson does know quite a bit about music, and that makes him half qualified for the job, even in my jaundiced opinion. But his knowledge of audio electronics, electroacoustics, and psychophysics—forget it. It's an embarrassment to an otherwise outstanding journal.

Actually, Neil Levenson's bimonthly column in *Fanfare*, "New for Audiophiles," could just as well be appearing in *The Absolute Sound*, of which he is an alumnus. Like so many of the equipment reviewers from Harry Pearson's stable, he blithely inserts a new piece of equipment into his system, plays an old familiar recording (say, a CD transfer of some 78's from the 1930's), forms an instant opinion of the sound, and attributes the qualities he hears (or claims to hear) to the performance of the new equipment. Just like that, I kid you not. He makes absolutely no attempt to standardize his setup and his methods in order to achieve any kind of consistency or repeatability. It never occurs to him that maybe he is hearing something other than the effect of the device under test. Double-blind comparisons at matched levels? He probably thinks that's an event at the Special Olympics.

What irks me in particular is his smug confidence in the validity of his impressionistic evaluations and solipsistic *aperçus*. For example, in the May/June 1990 issue, he talks about the Harman/Kardon HD7500 CD player's "rhythmic deadness which made the bass line seem remote in time." And "perhaps because the HK cannot keep the bass in time with rest of the music, the mood of the performance was cauterized," he writes. "After about five minutes I was bored." Joel Flegler, how can you tolerate such untutored trash in the pages of your fine magazine? There exists no mechanism known to physicists that could make the bass go out of sync with the rest of the music in the HD7500. Neil Levenson appears to be so in love with the first little conceit that pops into his mind while he listens that he never gives it a second thought before putting it in his column.

I could go on with example after example of Levensonian howlers, but here's one that neatly demonstrates why he shouldn't be writing about electronics. In his review of the Sansui AU-X911DG integrated amplifier (July/August 1990), he writes: "I did not test or audition the built-in digital-to-analog converter. There are four digital inputs, one optical. I noted that the non-optical digital inputs are via the usual RCA-style jacks. This struck me as possibly not apt, because the impedance of RCA-style jacks is around a couple of hundred ohms whereas the owner's manual says to employ '75-ohm digital connection cable.' It seems to me that an RCA-style plug does not properly terminate a supposed '75-ohm' cable."

I couldn't believe my eyes when I read that. Here's a man who accepts money to write a semitechnical column for audiophiles, and he doesn't know (1) that those coax digital inputs are terminated with 75-ohm resistors and (2) that it's utterly meaningless to talk about the characteristic impedance of a half-inch long conduit such as an RCA jack unless you're well into the gigahertz band. Some expert!

Maybe the solution is to enroll Neil Levenson in one of those correspondence courses in electronics and ground him at *Fanfare* until he gets his mail-order diploma. ◊

Seminar 1989:

Exploring the Current Best Thinking on Audio
(Part III of the Continuing Transcript)

The complete transcript turned out to be much more space-consuming than we had anticipated, so we are reluctantly parceling out the rest of it in short installments to make room for more up-to-date matters.

The first two installments of our seminar have created a veritable fan club among our readers; this group of audiophiles would like the transcript to go on forever, and they apparently go to it first, before anything else in our pages. Others have told us that they want equipment reviews, not all this loosey-goosey rapping. To us, the decisive factor is always the possibility of increased knowledge, regardless of the editorial framework, but we concede that after two issues our mainstream concerns should no longer take a backseat.

To understand and enjoy to the fullest these shorter Parts III, IV, etc. of the seminar, make sure you have read Parts I and II, including the capsule bios of the participants.

(Here we continue exactly where we left off, in the middle of a discussion of binaural sensitivity to very small time offsets, on page 51 of Issue No. 14.)

LIPSHITZ: So there's no contradiction in that statement. And I think the point is, once the rise time of the system—or, said another way, once the bandwidth of the system—sufficiently exceeds the rise time or bandwidth of the hearing system, there is no point in having it any faster because it doesn't substantially or significantly alter what gets through. It's rather like, once the signal-to-noise ratio of your recording system is 10 dB better than the signal you want to record, it degrades the signal by such a smidgen of a decibel that another 30 dB of signal-to-noise ratio will not get you a detectable change in the signal-to-noise of the program. It's a comparable sort of situation. There's no point in it.

CLARK: I think these arguments come from people who are trying to defend a belief that they have an emotional stake in, and they search around for anything that looks like technical support and just advance it. I don't think there's any logic behind it.

EARGLE: Dave, if you look at the amplifier business for the last 20 or 30 years, everybody has been advertising that they could pass a square wave with a fundamental frequency of 20 kHz, and they were damned and determined to show you that the thing retained a square wave shape all the way up, without any overshoot or with minimal rounding or something like that. And they were the first ones who said that it's got to be this good in order to work in the passband of the ear.

LIPSHITZ: Which was fairly easy to do with amplifiers; you just make their bandwidths up in the hundreds of kHz range. However, with digital systems to get that bandwidth is a little bit excessive!

EARGLE: And I don't know of anybody today who bandlimits an amplifier to 20 kHz or anything near it.

LIPSHITZ: No, and that's why digital systems have come in for so much criticism because the transient signals, the square waves, don't look the way people are used to seeing them when they go through.

EARGLE: They look like sine waves. They *are* sine waves.

LIPSHITZ: But, you know, an experiment which we frequently do with students as a demonstration—I think that's on that ASA disc. Take a square wave of a fundamental frequency above 7 kHz so that—we believe that the true square wave is only odd harmonics—the third harmonic will be above 20 kHz; you can't hear it or any of the higher ones. So there should be no difference between that square wave and a sine wave of the same fundamental amplitude. Now the fundamental of a square wave is not the same in peak-to-peak amplitude as the square wave is; it's bigger. I think the ratio is $4/\pi$. That's 4 over 3-point-something, so it's a bit more than 1.

EDITOR: The amplitude 4 representing the sine wave and the amplitude 3.14 representing the square wave?

LIPSHITZ: Yes. So, if the square wave was 1 volt peak to peak, the sine wave would have to be $4/\pi$ volts peak to peak. A bit more than one, if I remember the number correctly. I may be wrong; I think it's that. And you've got to watch most function generators—when you switch square to sine, it keeps the peak amplitude the same, not the fundmental. So you will hear a difference solely on the basis of the level change. And if you do this experiment properly, people don't believe it. They look at it on the oscilloscope: "I can't hear the bloody difference!" There's nothing that will convince people more rapidly than something like that...

CARVER: But they'll go away thinking that perhaps you've tricked them.

LIPSHITZ: ...but they don't do that experiment! And when you ask about these rise times and so on—put the square wave through. Use a very good system for your transduction. If you could make one with a bandwidth of 100 kHz, you can have a rise time down in the microsecond range. So that means you're letting the first umpteen harmonics of the square wave into the ear—you still won't hear the difference, unless you make the level so gross that you actually are creating significant intermodulation nonlinearities in the ear.

CARVER: And that's the only way you can hear 20 kHz. Because for somebody my age to hear 20 kHz requires a sound pressure power about three trillion times greater than to hear 1 kHz, and at 15.750 kHz, the television oscillator, it requires about a million times more sound power to hear than at 1 kHz.

LIPSHITZ: There is a benefit in not hearing the 15.750, mind you, isn't there? It helps not to hear that.

CARVER: Well, I can still hear that.

LIPSHITZ: Oh, you can? I can't. I can't hear that, either.

CARVER: Even a perfect ear—it's just marginal. You're only about 10 dB above the threshold.

EARGLE: You remember the London convention—oh, about seven or eight years ago—when KEF put on that demonstration of the audibility of various things?

LIPSHITZ: Clipping and that sort of thing?

EARGLE: Yes. And one test was the audibility of a noise signal extending beyond 20 kHz. It was a double-blind test.

LIPSHITZ: Oh yes. Yes.

EARGLE: And Laurie Fincham had come up with some sort of a ribbon tweeter, a

Japanese ribbon tweeter, that he mounted on these KEF speakers, so that the speakers actually went up that high—because his own speakers wouldn't do it. Anyhow, only one person out of the entire population statistically beat the odds, and it was an 18-year old guy from Denmark. He was the only one.
LIPSHITZ: Where was the cutoff of the filter for comparing?
EARGLE: Something like 23 or 24 kHz.
LIPSHITZ: Versus?
EARGLE: Twenty.
LIPSHITZ: Versus 20 kHz?
EARGLE: Yes. And he is the only one who could really hear it. Now, when you go from 20 down to 15, lots of people heard it.
LIPSHITZ: Oh yes, absolutely.
EARGLE: The idea being—we all have to ask ourselves this question—if one person out of a population of maybe, oh, let's say 250 members of the area who might have volunteered for these tests, if one person heard bandwidth in excess of 20 kHz, does that make it worth pursuing, intellectually, as engineers, or commercially? Probably no.
EDITOR: Don't say that too quickly.
EARGLE: Okay, I won't. But I think it's a point worth elaborating for the sake of our discussion here. What are our obligations? How do you feel about it?
CARVER: You're right. If it were possible to hear 22 or 23 kHz...
EARGLE: Yes, if those people could hear it, should we make records for them?
CARVER: ...if it were possible, and if they did hear it, I mean the SPL's must be just incredible at those frequencies, and it can't do anything but be hurting something, I would think.
EARGLE: No, it isn't that...
CARVER: I mean, you're going to be around at least 100 dB if you're going to hear 22 kHz...
LIPSHITZ: No, not necessarily.
EARGLE: No, at that age not at all.
LIPSHITZ: It depends. Some people do go beyond 20 kHz without enormous troubles.
EDITOR: Wouldn't you say that a perfectionist sound system should be designed for the most perfect ear among us?
LIPSHITZ: How do we find that person? We first have to find that person and measure him or her before we can design the sound system then, Peter. So we have a bit of a problem. How many—3 billion people in the world now?
EDITOR: I could live with the most perfect one found so far.
LIPSHITZ: We better check them all quickly before we go any further.
EDITOR: "Found so far" I think is good enough.
CARVER: I've checked a lot of hearing and I've only found one person who can hear even 21 kHz, and again it was a very young person and everything was perfect. And it required a tremendous sound pressure before he would even hear it. I think I was overloading something.
LIPSHITZ: And in music, there is no musical content up there. Even if at a point you detected it...
EARGLE: Well, in normal music... I don't know. I don't know. When you talk about synthesizers...
LIPSHITZ: Well, if you know a composer who is composing for that one person...
EARGLE: Look, Wendy Carlos can make an arrangement of "Switched-On Bach" with an oscillator going all the way up to 25 kHz. Should it be heard or should it not be heard?
LIPSHITZ: Well, is she doing that because there is somebody who *can* hear up there and she wants him or her to hear it?
EARGLE: Well, let's assume so.
LIPSHITZ: Or just because she didn't have a suitable filter to put at the output of the oscillator?
EARGLE: Let's assume that she is saying, if it can be heard, I'd like it to be heard. It's there to be heard.
LIPSHITZ: All right, I won't answer that. I'll answer it with a question.
EARGLE: Okay. That's fair.
LIPSHITZ: Suppose Wendy Carlos was not being recorded and released on records. Would you feel there's an obligation to record and release her work so that the one person who might be interested in hearing it can hear it?
EARGLE: Well, on the other hand, let me ask *you* a question. Let's say this is only on one piece and it's recorded as Band 1 on Side 1 of an LP...
LIPSHITZ: And that's not intended in any

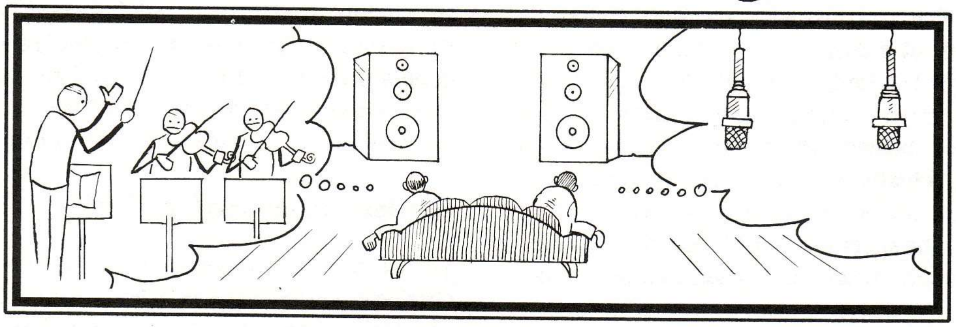
"...has it been an arbitrary decision...that 20 or 21 kHz ought to be the upper limit—what we're going to be recording from now until the end of time?"

way as a reflection on Wendy Carlos. This is just composer X.
EARGLE: Okay. Let's say this tone is recorded on Band 1 of one side of this LP, and you know it can be handled by that diameter. You can record to 35 kHz on the outer bands of an LP.
LIPSHITZ: Should you do so?
EARGLE: Well, you can play it back. It's there.
LIPSHITZ: Suppose you could record the RFI that your microphone line is picking up, *without* demodulating it, so that when you played your record back you could also get the FM broadcast that was going on in the background...
EARGLE: That would be a very wide-range system...
LIPSHITZ: ...as a choice, instead of being forced to listen to the demodulated version. *(Everybody finds this hilarious.)* For example, take Glenn Gould's Goldberg Variations, on CBS...
EARGLE: You hear a lot of musical externals going on.
LIPSHITZ: Yes, you hear plenty of Glenn Gould. But please tell me what the orchestra is that's playing in the background.
CLARK: Is there one?
LIPSHITZ: Oh yes.
EARGLE: Really?
LIPSHITZ: Yes. And I've had friends spend quite a few minutes trying to identify it. Because I wanted to write to CBS saying, "Why is Tchaikovsky's 5th playing in the background?" But we could not identify what it was.
CLARK: "And did you pay the royalties?"
EARGLE: Is that the early recording or the new one?
LIPSHITZ: No, no, no. The new one. The one recorded at CBS studios in New York City.
McGRATH: The last Goldberg?
LIPSHITZ: If you want to try it, everybody's got it. Yes, that one. Go to around 24 minutes.
EDITOR: I own that CD.
LIPSHITZ: Everybody owns it, man. CBS must be making a very good return on that.
EARGLE: Incredible.
EDITOR: I never noticed what you did.
LIPSHITZ: Listen around 24 minutes in the thing—that happens to be a good place to listen because the music's very quiet there—and you will hear this orchestral thing going on in the background. Now it's throughout the disc; it comes and goes. It's almost certainly RFI; you will hear that between variations it fades down and comes back up, so you know they digitally edited the thing. Fade it down, bring it back up...
McGRATH: If it had been done in analog, I would have thought maybe they didn't quite degauss the tape; I've heard that on other things.
LIPSHITZ: It's on the master tape; almost certainly it must be coming in on their mike lines; it's RFI...
EDITOR: The next thing would have been "Breaker two! Breaker two!"
McGRATH: "Good buddy Glenn, how ya doin'? Come back, good buddy Glenn." *(Laughter.)*
EDITOR: That has happened to me, on my system, in the middle of the music.
LIPSHITZ: ...but if those mike preamps and the digital system had been processed linearly...that I can hear it...demodulate the radio station WQXR or whatever it was, then... *[everybody talking at once, making this undecipherable—Ed.]*
EARGLE: Okay. I see the point you're aiming at here. And it isn't quite what I'm talking about.
LIPSHITZ: I know, I know. *(Laughs.)*
EARGLE: The thing is that we have cartridges and LP's and tape machines, at 30 ips, that will handle well beyond 20 kHz. It's been shown that some people can hear some distance beyond 20 kHz. The question then becomes, basically, has it been an arbitrary decision, or why have we decided, that 20 or 21 kHz ought to be the upper limit—what we're going to be recording from now until the end of time?
CARVER: Wait a minute. Wait, wait. It's been shown that some people can hear beyond 20 kHz *but* at such horrendous SPL's that it will never show up at that level. And if there's a 21 or 22 kHz *musical* bit on our recording, even that person is not going to hear it; it will be below his threshold at 22 kHz; he just won't hear it.
LIPSHITZ: But I appreciate John's point. I think my answer to you would be this. If there were no good engineering reasons for

wanting to restrict the bandwidth to something round 20 kHz, I'm sure we would have it wider.
CLARK: Just like amplifiers.
LIPSHITZ: Not because there are some people but just to give us a bit of extra leeway. Not because I happen to know one individual who could hear it. But the penalty, the engineering penalties...
EARGLE: It's an economic penalty.
LIPSHITZ: ...for increasing the bandwidth from 20 to 25 kHz in digital are extremely high. And given the choice that that might mean giving you 25% less playing time on your CD, for that one individual, I doubt that many people would make that decision.
EARGLE: Okay. I think you're probably absolutely right and I would say that it becomes a matter of the cost of real estate in the medium. And I would expect to see this particular point made in the transcript of this meeting we're having today. *[Come on, John, every word is being transcribed; that's why the damn thing is so long.—Ed.]*
LIPSHITZ: It's an engineering trade-off.
EARGLE: It is a trade-off. We would not choose to do it for any other reason.
McGRATH: But is that the reason it was chosen in the first place? I mean, why did Philips settle on the bandwidth that they did?
LIPSHITZ: Oh, there's a very good reason for that.
CLARK: That's an even number, and twice that is 44.1, which they already have...
LIPSHITZ: No. No. The reason is the following. The only feasible way of recording digital was on video recorders. If they hadn't established a format for recording on video recorders, there would have been no digital material available at the time of the CD release.
EARGLE: Soundstream was generating all the 50 kHz material, and there are a lot tapes around which have been transcoded over to 44.1.
LIPSHITZ: Right. But the question was, why was the CD 44.1. My answer is that the only practical standard that they could come up with—we're talking now...going on to a decade ago—was based on the video format, and that means you had to have an integral number of samples per horizontal line, and that led to 44.056 kHz in NTSC 59.94 fields-per-second countries—it would be 44.1 for 60—but anyhow, that's what led to that number. So it's just de facto; if there had been other ways of storing it, I'm sure a much more attractive-looking number than 44,100 would have been chosen.
EARGLE: Yes, and you know, when you look at the entire standard, I've never been put off by the 16-bit limitation. That's never bothered me at all. But I have sort of wondered from time to time about the sampling rate because in our time, in this business, we've seen it: begin at 50, with Soundstream; we've seen it go to 48; and then we've seen it drop precipitously down to 44.1. One can only look at that and say, well, you better watch those bastards or they're liable to lower it again. It isn't going to happen, but for a while...
LIPSHITZ: There's a theorem that will prevent them from lowering it anymore.
EARGLE: That's right.
EDITOR: It's like the fear of rising taxes—you have a fear of lowered sampling rates. *(Laughter.)*
LIPSHITZ: But it's interesting... The funny thing is, John, you're happy to accept 16 bits, but you're reluctant to have too low a sampling rate, too close to the Nyquist limit...
EARGLE: Yeah. Yeah.
LIPSHITZ: ...yet, of those two trade-offs or engineering decisions, the former is the questionable one, and the latter is the ironclad one, in principle.
CLARK: Yes, that's right. That's right.
LIPSHITZ: If you can't hear above 20 kHz, you don't need a sampling rate more than two times that, plus whatever guardband you need for reasonable filter design.
EARGLE: Of course.
LIPSHITZ: Whereas the decision to use any finite number of bits, in principle, is a degradation, a loss. We can change what would be a distortion into a noise, by dithering or doing other things, but that's what fixes your signal-to-noise ratio; it is a trade-off. The noise can be heard. So you've introduced a noise, which in an infinite-bit system would not have been there.
EARGLE: Well, it's a noise which you can argue about, in terms of the dynamic range of music. The thing is that if you measure

"My point is that, had consciences not prevailed here, we could have ended up with a medium that was 15-kHz limited and 14-bit limited."

in any concert hall the maximum level of an orchestra and then look at the noise level into which it sinks when the music stops, you're nowhere near the inherent dynamic range limitation of a 16-bit system.
LIPSHITZ: True, but you're nowhere close to what the ear is capable of.
EARGLE: That's right.
CARVER: So when we combine what *you* said with what *you* said, we find that the existing digital system very comfortably fits totally inside the envelope of the human hearing mechanism, without being troubled at all.
EARGLE: Yes, it does.
McGRATH: No, no, no—the music replication mechanism, not the human ear.
LIPSHITZ: The music replication, yes...
McGRATH: The ear hears much lower than the 16 bit...
CARVER: Oh, yes, yes.
McGRATH: ...but the real world does not demand more than 16 bits.
LIPSHITZ: ...what I'm saying is, the 16-bit decision has introduced a noise floor which in realistic music, as John is saying, is not a limitation. He can't find a hall that's quieter to record in. However, if I wanted to have a recording system that would *never* add its own noise to any recording I made, I need a dynamic range of at least 120 dB—because that's the approximate dynamic range of the ear. And that means that I need more than 16 bits. Where does that take us up to...? 18, 20... We need about 20 bits to do that. And then you get very dicey, if you need more than *that*. Because then you're designing for creatures we haven't found yet, who maybe will visit us from outer space one day—and you wouldn't want to limit their dynamic range... *(Laughter.)*
CLARK: Well, we should have some music for them...
EARGLE: I suggest then that we further state that the decision to use 16 bits is in itself an unfortunate compromise and say that we around this table all wish that these limits could be extended.
LIPSHITZ: No, I'm not sure I wish that one.
CLARK: I don't wish to pay for it.
EARGLE: Well, okay. It sound to me as though everybody here is in agreement that Philips or Sony—and I don't trust those bastards any further than I could throw them *(spoken with a facetious inflection)*—made the right compromise for mankind forevermore. Now, I'm sure you don't think that.
LIPSHITZ: Now wait. We've already discussed the sampling rate, and I think we said, all else being equal the engineers would probably have chosen a higher one if they'd been able to *(inaudible word)* for it.
CARVER: For comfort reasons...
EARGLE: And they probably would have chosen more than 16 bits if they had the leave to...
CARVER: ...for just sort of comfort reasons, but not for any real demonstrable reasons.
LIPSHITZ: Well, look. The original Philips conception was 14 bits. It was as a result of Sony's prodding that it was raised to 16—because it didn't require unacceptable loss of playback time on the medium, thanks to clever channel encoding and so on.
EARGLE: I must say something here. I wanted to talk about unfortunate compromises.
LIPSHITZ: Yes, let him modify his statement, then I'll comment on it.
EARGLE: I won't modify it; I'll elaborate. I'm going to say that if we had had Philips succeeding with their 14-bit suggestion here, and somebody else were saying that, well, FM, which our best medium in many ways, is limited to 15 kHz, so why not go to 32 kHz sampling—and where do you think that unfortunate number came from?
LIPSHITZ: No, *that* I call unfortunate.
EARGLE: Okay.
LIPSHITZ: 14-bit I would call unfortunate because it's not significantly better than analog tape. 16-bit I don't call unfortunate.
EARGLE: Okay. My point is that, had consciences not prevailed here, we could have ended up with a medium that was 15-kHz limited and 14-bit limited.
LIPSHITZ: Yes.
CARVER: That's true. We came close to that.
LIPSHITZ: We came close.
CARVER: We came close.
LIPHITZ: But it's like the slew-rate ques-
(continued on page 68)



In the immortal words of Julius, jacta alea est, *the die is cast. (No, not Julius Futterman, audio freaks. Julius Caesar.) This will be strictly a CD and DAT column. There will be no LP's reviewed here, even though that possibility was brought up three issues ago. LP's are no longer a factor in purchasing decisions involving new releases, and our precious vinyl heritage is currently being archived on CD with considerably greater skill than was the case initially. A widely traveled industry figure recently observed that various audio journalists known for their antidigital protestations seem to be playing nothing but CD's these days; their turntables are always being serviced, updated, exchanged, borrowed by a friend, etc., and therefore not available at the moment. Very interesting—and not surprising.*

CD's from the Golden Age of Audio (Meaning Right Now)

By Peter Aczel
Editor and Publisher

We have reached the point in the evolution of the CD where the least good new release sounds pretty nice and the best sounds awesome—and the same is true of the new CD players. I call that the golden age of audio because LP's and phono cartridges never came close to giving that kind of consistent satisfaction, even if the absolute best of them were quite wonderful. Today the medium is on an entirely new level of sonic reliability.

For that reason I want to get away from grouping my reviews label by label, as if that were the major determinant of audiophile interest. Although a few labels still offer more predictably excellent sound than others (especially the three D's and the double R—Delos, Denon, Dorian, and Reference Recordings), surprisingly good engineering is currently encountered with some regularity on many labels, major and minor. The reasonable thing to do now is to alphabetize by composer, like the catalogs, and treat the sound as just one more attribute of the recording.

Beethoven

Ludwig van Beethoven: Piano Sonatas. John O'Conor, piano. Volume I: Sonata No. 8 in C Minor, Op. 13 ("Pathétique"); No. 14 in C-sharp Minor, Op. 27, No. 2 ("Moonlight"); No. 23 in F Minor Op. 57 ("Appassionata"). Volume II: Sonata No. 21 in C Major, Op. 53 ("Waldstein"); No. 17 in D Minor Op. 31, No. 2 ("Tempest"); No. 26 in E-flat Major, Op. 81a ("Les Adieux"). Volume III: Sonata No. 15 in D Major, Op. 28 ("Pastoral"); No. 16 in G Major, Op. 31, No.1; No. 18 in E-flat Major, Op. 31, No. 3 ("Hunt"). Volume IV: Sonata No. 1 in F Minor, Op. 2; No. 2 in A Major, Op. 2; No. 3 in C Major, Op. 2. **Telarc** *CD-80118, CD-80160, CD-80185, CD-80214, respectively (all DDD, recorded 1985/1989 by Jack Renner, released 1986/1990).*

Prof. Charles Rosen of the University of Chicago, one of the outstanding intellects of the music world and something of a cult figure as a pianist, points out in a recent article in *The New York Review of Books* that the piano sonatas of Beethoven and Schubert were not composed for concert-hall performance but, in most cases, for a semiprivate *soirée* audience of 20 or 30 Viennese music lovers. John O'Conor plays Beethoven as if that were his principal guideline, and I find his intimate style, with its total eschewal of declamatory distensions, quite refreshing. My feeling as I listen to him is that he is playing for *me*, not for an adoring gallery, and that he wants me to understand the structure and flow of the music. Not that he is lacking in virtuosity; he is fleet-fingered and accurate enough to satisfy any nitpicker of key-board technique; but his aim is to make his audience say "aha" or "hm" rather than "wow." One could, of course, cynically speculate that a pianist who possesses neither the emotional profundity of a Schnabel nor the brilliance of a Horowitz is probably reduced, willy-nilly, to such a "good-taste," musicianly approach. I feel, however, that a complete set of the Beethoven sonatas, which this Telarc series will eventually become, actually gains a certain reference value by avoiding extremes and normalizing its performance rhetoric to the middle of the spectrum. That helps to bring hotter and colder performances, when they occur, into proper perspective.

Since the highest opus number in these four volumes is 81a, it remains to be seen, or rather heard, how the O'Conor treatment suits the biggies like Op. 106 ("Hammerklavier") or Op. 111. His "Appassionata" is already a bit tame (i.e., unimpassioned) for my taste, although quite lovely in many respects. Hey, maybe his taste is more refined than mine.

As for the audio quality of these four CD's, Volumes III and IV were recorded in the same hall in Worcester, Massachusetts, with identical equipment, whereas Volumes I and II represent slightly earlier implementations of the same technique in two different halls in England, so that the sound is similar throughout but perhaps slightly superior in the later recordings. A spaced pair of Brüel & Kjær 4006 omnidirectional condenser mikes is the common denominator of them all; like the playing of John O'Conor, that constitutes another "normative" factor here—even a recording engineer as pragmatic and as different from Jack Renner as John Eargle uses the very similar 4004's for solo piano—resulting in a solid, well-delineated, dynamic, completely unproblematic piano sound with a fairly close-up perspective but without any blow-your-socks-off ambitions. I can recommend this continuing series by the rising young Irish artist to all those who like their Beethoven straight up.

Berlioz

Hector Berlioz: Symphonie Fantastique, Op. 14. Frankfurt Radio Symphony Orchestra, Eliahu Inbal, conductor. ***Denon*** *CO-73208 (DDD, recorded 1987 by Detlev Kittler, released 1989).*

The most fantastic thing about Berlioz's fantastic "episode from the life of an artist" is that it was composed in 1830, only six years after Beethoven's Ninth. The stylistic light-year Berlioz was able to put between himself and all who had preceded him still doesn't fail to astonish. This a very fine performance of this amazing work, in the deliberate, meticulously constructive manner of Inbal, who obviously believes that too many climaxes would be equivalent to no climax at all. He graduates the tension from movement to movement, phrasing each measure with exactly the emphasis it organically needs, and at the end the work escalates to an overwhelming blaze of glorious sound. There is something to be said for taking your time.

The recording with Denon's usual B&K technique in the Alte Oper in Frankfurt is very successful; the violins are smooth as silk; the timpani, bass drum, and lower strings are audible in almost frightening detail; but what are those RF birdies doing on Track 3, beginning at 4:41? This is too good a production for something like that to be allowed to slip through. Even so, if you believe in owning more than just one or two *Fantastiques*, get this disc.

Bernstein/Barber/Gershwin

Samuel Barber: Overture to "The School for Scandal." George Gershwin: An American in Paris. Leonard Bernstein: Arias and Barcarolles. Seattle Symphony, Gerard Schwarz, conductor; Jane Bunnell, mezzo-soprano; Dale Duesing, baritone. ***Delos*** *DE 3078 (DDD, recorded 1989/1990 by John Eargle).*

Along with the Walter Piston album reviewed below, this CD represents a new standard of sonic excellence in the Schwarz/Seattle series engineered by John Eargle, which I have already glorified more than sufficiently. The sound is cleaner, crisper, more transparent than ever, without even a suggestion of strain or hardness. Definitely 1990 demo quality. Could it be an improved method of digital encoding? The credits cryptically list "Digital Recording: Sony" where there used to be model numbers of the digital processing equipment. Very interesting...

A little over 50% of the music is a Bernstein joke, a cutesy farrago of fluff and nonsense composed (concocted?) in 1988, which Lenny must have considered very important because he let one Bright Sheng do the orchestration. I am decidedly underwhelmed. The best reason to buy this disc is a highly idiomatic performance of Gershwin's "An American in Paris" in—get this!—the uncut, original version that apparently has never been performed until now. About three minutes of music already fully orchestrated by the composer were excised from the premiere performance, for reasons no longer known, and can be heard here for the first time. All that and a great bass drum, too, in the familiar passages. Samuel Barber's eight-minute graduation thesis from the Curtis Institute (he was 22) is also a class act, a very nicely crafted piece of 1930's eclecticism and very nicely played.

Although I wouldn't include this album in "All the Classical Music Your Family Will Ever Need" (remember that incredible TV commercial?), I can recommend it for half of the music and all of the sound.

Bruckner

Anton Bruckner: Symphony No. 4 in E-flat Major ("Romantic"), ed. Nowak. Royal Concertgebouw Orchestra, Riccardo Chailly, conductor. ***London*** *425 613-2 (DDD, recorded 1988 by Colin Moorfoot, released 1990).*
Anton Bruckner: Symphony No. 4 in E-flat Major ("Romantic"), original 1874 version. Cincinnati Symphony Orchestra, Jesús López-Cobos, conductor. ***Telarc*** *CD-80244 (DDD, recorded 1990 by Jack Renner).*

I happen to be one of those who find much of Bruckner's music disjointed, naively grandiose, and lacking the natural fluency of other grandiose 19th century composers like Wagner, Liszt, or Tchaikovsky. Nevertheless, I concur with the musicologist Alfred Einstein that Bruckner "produced his most harmonious work in his Fourth Symphony, which depends almost entirely on beauty of sound."

For beauty of sound you would expect the Royal Concertgebouw to be an easy choice over the Cincinnati Symphony, but such is not the case. López-Cobos appears to have much more of a *con amore* approach to the symphony than Chailly, or perhaps more of a sense of occasion, since he is conducting the very rarely performed 1874 *Urtext* of the Fourth, published only in 1975, the first of five different versions by the perpetually self-doubting composer. The Cincinnatians respond to their conductor with beautiful sonorities and great discipline, and Jack Renner's recorded sound is more beautiful than Colin Moorfoot's, although the latter is very respectable in the Decca/London multimiked idiom. Jack Renner is back to his trusty old Schoeps omnis and getting much sweeter upper midrange and lower treble than in his recent Sennheiser recordings.

I unhesitatingly choose the Chailly, however, as the musically preferable disc of the two. The original 1874 version has a Scherzo totally different from and not nearly as brilliant as the "Hunting Scherzo" of the later versions, which also have a considerably altered Finale. As a non-Brucknerian, I always thought it was those amazing brass passages in the later Scherzo that made the Fourth worth the price of admission. In this case Bruckner wasn't just being insecure to have listened to criticism; he did improve on the symphony, at least in my not very authoritative opinion. Therefore, historical/puristic considerations aside, I opt for the standard Nowak edition—but then Chailly has lots of competition from greater conductors.

Dvorák

Antonín Dvorák: Piano Quartet in D Major, Op. 23; Piano Quartet in E-flat Major, Op. 87. The Ames Piano Quartet: Mahlon Darlington, violin; Laurence Burkhalter, viola; George Work, cello; William David, piano. ***Dorian*** *DOR-90125 (DDD, recorded 1989 by Craig Dory, released 1990).*

For once I have to criticize, albeit mildly, Craig Dory's work at his favorite recording site, the Troy Savings Bank Music Hall. This recording of three stringed instruments and a piano is too reverberant; it should have been miked a little more dryly and intimately. I think Craig is so much in love with the unique acoustics of the TSBMH that he wants to make sure you don't miss the slightest nuance of it, but in this case the frame, so to speak, threatens to overwhelm the picture. Don't misunderstand me; it's still a very nice, musical sound, but from Craig I expect a "ten" every time.

The music here is one early work of Dvorák and one mature spellbinder; the players are excellent but not world-class; even so they play the exceptionally lovely Lento of Op. 87 with sufficient flair to make me almost like the not-quite-right recording. That movement, excerpted in advance on a Dorian sampler, remains the high point of the album. Overall, an honorable near miss.

Elgar

Sir Edward Elgar: Cockaigne (In London Town), Concert Overture, Op. 40; Variations on an Original Theme (Enigma Variations), Op. 36; Serenade in E Minor for Strings, Op. 20; Salut d'amour (Liebesgruß), Op. 12. Baltimore Symphony Orchestra, David Zinman, conductor. ***Telarc*** *CD-80192 (DDD, recorded 1989 by Jack Renner, released 1989/1990).*

The first thing that struck me when I began to listen to this was how well the Baltimore orchestra was playing. No provincial symphony, this one. David Zinman is obviously doing a good job. The recording, too, shows Jack Renner in very good form; he seems to have figured out the Joseph Meyerhoff Symphony Hall in Baltimore to a T, and here again he is back to his nice Schoeps omnis after his flirtation with those overly aggressive Sennheisers. This is the kind of sound I expect from Telarc (see Issue No. 14, p. 39, third column).

The music is of course familiar; I don't claim to be an Elgar maven but I find his anglicized late-19th-century idiom without sentimentality quite exhilarating. The "Enigma Variations" are definitely a minor masterpiece—maybe not even so minor—and the "Cockaigne" overture is rousing good fun. Zinman conducts all of it with authority and sensitivity; short of a DDD recording of a resurrected Toscanini or Sir Adrian Boult, this will do.

Franck

César Franck: Symphony in D Minor; Variations symphoniques. Royal Philharmonic Orchestra, Claus Peter Flor, conductor; Rudolf Firkusny, piano. ***RCA Victor Red Seal*** *60146-2-RC (DDD, recorded 1988/1989 by Mike Hatch & Mark Vigars, released 1990).*

To me, the important part of this release is the Variations for piano and orchestra, not the Symphony. I like only the second movement of the latter; the rest strikes me as a bunch of overstated platitudes. Thus I am disinclined to judge the finer points of what is an obviously well-played, musicianly performance. The Variations, on the other hand, are a lovely, beautifully crafted, classically restrained work of lasting appeal—and an excellent vehicle for one of my longtime heroes, Rudolf Firkusny. He is now 78 years old, but when he was 50 he absolutely *owned* the Beethoven Third Piano Concerto, for example (in my opinion, at least), and his Mozart was equally marvelous, not to mention the Czech masters, which of course he played in the best native idiom. Here he is still the aristocratic, singing-toned, lyrical player of old, and the 15-minute piece becomes an intense musical experience under his hands. The 37-year old German conductor, Flor, is new to me, but he is as good as is needed for this effort and so is the Royal Philharmonic.

The RCA recording is interesting because it is so smooth, sweet, rounded, and trouble-free. I suspect a commercially calculated restriction of dynamic range, making the album sound good on just about any system but not extraordinarily good on systems such as mine. Anyway, it's a better philosophy than boosting everything to the point of unpleasantness.

Handel

George Frideric Handel: Arias for Montagnana. David Thomas, bass; Philharmonia Baroque Orchestra, Nicholas McGegan, conductor. ***Harmonia Mundi*** *HMU 907016 (no SPARS code, recorded 1989 by Peter McGrath, released 1990).*

When I heard excerpts from this for the first time at the 1990 Summer CES, I went to the Harmonia Mundi booth, identified myself to the man in charge, and said, "I can't live without this CD." He took pity on me and gave me a copy for review. What I didn't know then was that there's a certain sameness to all these florid, virtuoso bass arias (if you've heard three, you've heard all 17), but taken a few at a time, they remain absolutely dazzling even after repeated exposure to them.

Antonio Montagnana was the number one bass of Handel's oratorio company in 1732 and 1733; he was famed for his range, intonation, and superaccurate leaps. Handel composed Montagnana's showstoppers specifically to display his vocal equipment; the tendency toward repetitiousness is therefore not surprising. A good thing bears some repetition, however, and this music is definitely a good thing.

The surprise here is the English bass, David Thomas, widely known as a highly competent and thoroughly musical baroque specialist, but not as a superstar. Here he is one. His stylistic confidence, vocal security, and general panache remind me of Ezio Pinza, although he may not quite match the latter in sheer beauty of voice. His bottom notes are astonishing; his energy is inexhaustible. You shouldn't deny yourself the experience of hearing this. What a singer! And, for once, I find myself in total agreement with McGegan's handling of the often brilliant orchestral parts—gorgeous playing, wonderful articulation.

The recording is something of a departure for Peter McGrath; the venue is a big sound studio at Lucasfilm in California, very different from his usual school chapels. I find the recorded sound to be a bit too reverberant for the music, but every note is crystal-clear and sweet, and the balances are perfect, so it really comes down to a matter of individual taste. I assume this is another Schoeps omni job, but why no SPARS code, Peter? Have you gone underground as an AAD diehard?

Mendelssohn/Schubert

Felix Mendelssohn-Bartholdy: Piano Trio No. 1 in D Minor, Op. 49. Franz Schubert: Piano Trio No. 1 in B-flat Major, D898. The Rembrandt Trio: Valerie Tryon, piano; Gerard Kantarjian, violin; Coenraad Bloemendal, cello. Dorian DOR-90130 (DDD, recorded 1989 by Craig Dory, released 1990).

This, too, is a Troy Savings Bank Music Hall chamber music recording, just like the Dvorák album above, produced by the same team only two months later. The results, however, are distinctly superior in my opinion; the sound is a little less reverberant, with still plenty of liveness and sufficient TSBMH signature, and the three instrumentalists are superb.

These are wonderful works, needless to say, by two of the greatest masters of the piano trio form, so there is very little else to say except perhaps that this is what today's golden age of audio is all about. There may be greater performances than these in the catalog (bristling with names like Casals, Heifetz, Rubinstein, etc.), but this combination of lovely playing and you-are-there sound is a pleasure unique to our era.

Piston

Walter Piston: Symphony No. 2; Sinfonietta; Symphony No. 6. Seattle Symphony (in the Symphonies); New York Chamber Symphony (in the Sinfonietta); Gerard Schwarz, conductor. ***Delos*** *DE 3074 (DDD, recorded 1988/1989 by John Eargle, released 1990).*

This is the CD John Eargle trots out these days when he wants to show his stuff to the fans. Enough said. The Symphony No. 6, especially, is state-of-the-art orchestral recording, in transparency, spatial perspective, dynamics, and freedom from even momentary nasties (it was the last to be taped). To use John's own terminology, both "texture" and "structure" are close to perfection. Again, the credits say "Digital Recording: Sony" without specifying the actual encoding equipment (see also the Bernstein/Barber/Gershwin review above), so I don't know whether that had anything to do with the new sonic heights scaled here. John claims it was simply the right orchestration matching the right hall on the right day; he didn't do anything special. Modest man.

I am quite thrilled by the Seattle/Schwarz revival of mid-20th-century American symphonists (Walter Piston, Howard Hanson, David Diamond, etc.) because it has made me more keenly aware of the ridiculous musical snobberies prevailing during my formative years. My generation was expected to delight in Schönberg, Webern, and Elliott Carter, not to mention Stravinsky's dodecaphonic dotage, while smiling condescendingly at the highly accessible works of those neoclassical and neoromantic eclectics. What nonsense! Walter Piston is more enjoyable than the twelve-tone contortionists and at least as good a craftsman. If you like Shostakovich (and even the old snobs allowed that, since Stalin had forbidden him to compose like Alban Berg, right?), then there's no reason why you shouldn't find Piston's leaner, more classically restrained, but just as "popular" idiom equally appealing. The orchestration alone is worth your careful attention. And the Seattle Symphony is getting better all the time. Keep 'em coming, Delos.

Ravel

"Nojima Plays Ravel." Maurice Ravel: Miroirs, Gaspard de la nuit. Minoru Nojima, piano. Reference Recordings RR-35CD (DDD, recorded 1989 by Keith Johnson, released 1990).

When I raved about Nojima's Liszt in Issue No. 12, I was reacting to his astonishing combination of technique, drama, and lyricism. The man has the "chops" to play anything accurately at any speed and any volume, but he is equally capable of expressive, singing phrasing when the music calls for it and doesn't feel the need to showcase his strength when that's not the point. Liszt gives him the opportunity for such contrasts of power and gentleness; Ravel is another matter. Ravel is cool, in the jazz sense. Rhythm, color, dynamic nuances, precision are the name of the game in Ravel, and it's a game that Nojima plays equally well. These are awesome performances.

This is supposed to be some of the hardest-to-play piano music in the world (especially the *Gaspard de la nuit*), but you wouldn't know it listening to Nojima. He is experiencing about as much difficulty as if he were playing *Für Elise*. The characteristic sonorities and coloristic effects of Ravel come through in his playing with greater conviction than I have heard from anyone since Gieseking, and the slower passages have continuity where lesser players tend to fragment them. And when flying fingers are needed—wow! At moments the piano whistles like the wind. Of course, Ravel is Ravel, not Beethoven. He doesn't transport you to Elysium (not me, anyway), and I don't expect the performer to make him sound transcendental. His music is a feast for the ear, not a probing of our emotions, and Nojima does it full justice.

The recording by Keith Johnson makes the upper strings of the piano, so important in this music, sound more vivid than I ever expected to hear through loudspeakers—absolutely stunning—but the hiss from the microphone electronics obtrudes even more than in the Nojima/Liszt album, probably because there are more pianissimo passages. I buttonholed Keith at the CES and asked him about this; the problem seems to be the very low output from the Coles figure-eight ribbon mike combined with less than ideal transformer matching to the preamp input. Keith has various fixes in the back of his mind but hasn't gotten around to them yet. To archive such a rare performance, I personally would have opted for a hiss-free B&K recording with possibly a smidgen less upper-string realism, but that's a value judgment which may not be Keith's or Tam Henderson's. Hiss or no—get this CD.

Schubert

*Franz Schubert: Mass No. 2 in G Major, D167; Mass No. 6 in E-flat Major, D950. Atlanta Symphony Orchestra & Choruses, Robert Shaw, conductor. In Mass No. 2: Dawn Upshaw, soprano; David Gordon, tenor; William Stone, baritone. In Mass No. 6: Benita Valente, soprano; Marietta Simpson, mezzo-soprano; Jon Humphrey, tenor; Glenn Siebert, tenor; Myron Myers, baritone. **Telarc** CD-80212 (DDD, recorded 1988/1989 by Jack Renner, released 1990).*

Go to Track 9, 2:28, on this CD. Listen to the *Et incarnatus est* in the *Credo* of the E-flat mass. Isn't this the most beautiful music in the world? And hardly anyone ever talks about it! I must admit, of course, that I became addicted to it on the basis of the Erich Leinsdorf recording from the 1960's (with the Berlin Philharmonic and St. Hedwig's Choir on EMI), not because of this performance. Leinsdorf's first tenor in that magical , sunlit nativity passage was the one and only Fritz Wunderlich; Shaw's soloists are not in that league—but who is? Shaw's choral work, on the other hand, is outstanding, maybe better than Leinsdorf's, and the many great choral passages come off splendidly (for example the brooding *Crucifixus* that alternates with the *Et incarnatus*). This is late Schubert, composed in the last year of his life, when he produced nothing but towering masterpieces. The early Mass is a simple, tuneful, and unimportant work.

As a footnote, I'd like to observe that Bach, in his B Minor Mass, treats the incarnation as a dark mystery already foreboding the pain and tragedy of the crucifixion, which is then followed by the unbridled joy of the resurrection—whereas to Schubert the incarnation is a pastoral idyll, the crucifixion a fateful disaster, and the resurrection almost a matter of course. The Protestant versus the Catholic perception, one could argue.

As for the sound, you can always assume that a Jack Renner choral recording will be excellent, and that is definitely the case here. Transparency, inner detail, spatial qualities, dynamics, transients are all what they're supposed to be at the summit of the art. Surprisingly, both Sennheiser and Schoeps microphones were used, so my theories about that are shaken, but then a choral/orchestral recording with solo singers has its own special rules. I love Schubert more than I love my theories, so I say— nice disc.

Verdi

*Giuseppe Verdi: Messa da Requiem. Oberlin Musical Union & Orchestra, Daniel Moe, conductor; Carolyn James, soprano; Susan Toth Shafer, mezzo-soprano; Franco Farina, tenor; Gerald Crawford, bass-baritone. **Bainbridge Records** BCD2103, Discs 1 and 2 (DDD, recorded 1988 by Brad Miller, released 1988).*

This not a very new recording, but I received it only recently and find it rather interesting. It documents a live, large-scale musical event—in celebration of the 100th anniversary of the Verdi Requiem's first performance at the Oberlin College Conservatory of Music—strictly from the perspective of the conductor. To quote Brad Miller's note: "Our intent was to archive an 'event' as it occurred; or to be more succinct, digitally record this 'live' performance from the perspective of Daniel Moe. A single microphone position was chosen, center stage and elevated over the podium.

The Colossus, a state-of-the-art 4-channel digital recording system, employed a direct feed from the producer's own MS-4 microphone. No outboard mixers or amplifiers were used. The MS-4 is a discrete 4-channel surround (quadraphonic) device, with a frequency response of 2 Hz to 20 kHz, ±2 dB. The microphone is DC servoed, which maintains low-frequency imaging very precisely, and will handle sound pressure levels (SPL) of 146 dB before clipping or distorting. The front and back channels were mixed together for this stereo compact disc..."

What do I think of the sound? Stupendous—but... The absence of any other microphone than the MS-4 makes the spatial perspective and overall balance utterly natural, uncomplicated, and plausible—the best I've heard. On the other hand, the words of the choir would have been much more distinctly audible with helper mikes, and I think any large commercial recording company would have opted for them, purism be damned. I'm not sorry, however; the disc as it is makes a wonderful imaging test tool. In every other way—dynamics, low-level decay, timbral accuracy, lack of distortion—the recording is fantastic. (No "ouch!" on the ladies' fortissimo top notes, not even a trace.) Since no subtle artfulness, as such, was possible in microphone deployment, I must attribute this success to the hardware—in which case other recording companies should take notice.

It has become something of a cliché that the Manzoni Requiem is operatic rather than religious in spirit, but what does that really mean? I think it means that Verdi always sounds like Verdi—but so does Beethoven, in which case the *Missa Solemnis* is symphonic, right? And isn't *Parsifal* religious in spirit even though it's an opera? I admit that the *Recordare* in the *Dies Irae* of the Requiem could be straight out of *Aida*, but that's something pretty good to be straight out of, as long as the singing is good. In this performance the singers are competent but not great, except the mezzo-soprano Susan Toth Shafer, who is superb. Her lower range, especially, is quite thrilling. The playing of the conservatory orchestra is on a respectable professional level without posing a threat to the ranking of their colleagues in nearby Cleveland. Daniel Moe paces and phrases the music effectively, but then anything by Verdi more or less plays itself; the "interpretation" is built in. (The man was a pro.)

Overall, I'd have to say that this recording makes more of an audio statement than a musical one, but it is far from inferior musically. ◊

Seminar 1989

(continued from page 62)

tion—isn't it?—with amplifiers. How close can you come to the slew rate before it's unacceptable...

CARVER: It's like falling off a cliff.

LIPSHITZ: ...how much margin do you need?

CARVER: As long as you don't fall off that cliff, you're perfectly okay. You can come as close as you please.

LIPSHITZ: There are people who believe you want a slew-rate margin of a factor of 10 so that you never come close to the limit. There are other people who believe that you can come up to 0.9 of the slew limit and not be able to tell that you're there.

EARGLE: And there are people who will tell you that a bridge ought to be built with 10 times safety factor in every area—and it's a prohibitively expensive bridge but it will never crash.

LIPSHITZ: But they never get the contract because their price is too high.

EARGLE: That's right.

LIPSHITZ: But anyhow, on this number-of-bits question, I would say the following. A 16-bit system gives a 98 dB signal-to-noise ratio in principle, undithered.

EDITOR: 98.1, right?

EARGLE: An even wider dynamic range.

LIPSHITZ: The dynamic range is wider because you can hear things below the noise. But that's true of any system. In practice, in a properly dithered system, you're talking more than 90 dB. Now, for most people, it is really quite eye-opening to play something very loudly, perhaps significantly beyond the level they would ever want to play music at, something corresponding to full modulation of the medium, and then play them a –90 dB tone—and they'll say, you haven't played anything; they'll say there's nothing on. That's what they'll say. Then you say, go to the loudspeaker, put your ear there. Oh yes, ah! You know, 90 decibels is really...

EARGLE: It's a lot.

LIPSHITZ: People don't quite appreciate it. Now, yes, the noise floor in a quiet room like my listening room is just at the level of the subliminal. You can perceive the noise floor of the digital system; it's not below my threshold. It would be nice if it were below; a few dB more might be nice. But, goodness, for a medium designed for hundreds of millions of people out there at an affordable price, it's not bad. It's pretty damn good. Now, the professional I think needs more than 16 bits because he needs to be able to do some manipulations and mixing and headroom and other things, and be able to get 16-bit signal-to-noise in the end.

EARGLE: Are you going to go on record with that statement?

LIPSHITZ: Yes.

EARGLE: Good.

CARVER: Now I understand what your objection was. And you just brought it out, Stanley. Because you need that headroom. You'd love to know that you're 12 dB away from clipping, or something like that, when you're operating your machines or editing your recordings. And the person who makes the final copy that we as consumers will listen to, he...

McGRATH: Will get the full 16.

CARVER: I suppose, I imagine that they somehow look through the thing, find the loudest passage, and park it right up at the top—and he has his 98 dB.

LIPSHITZ: Not just that, but multitrack digital.

CARVER: But when you're making the recording you don't have that luxury; you need the extra 10, 12, 15, maybe 20 dB, to make sure you never run out of headroom.

LIPSHITZ: You want some at the top, and it's also good to have some at the bottom, so when you're mixing twelve channels together, you don't get twelve 16-bit noise floors added together but twelve 18- or 20-bit noise floors.

CARVER: So that's where your 20-bit would be, yes.

CLARK: Essentially, there should be a difference between pro and consumer machines then.

CARVER: Yes.

EDITOR: What is the widest analog dynamic range that you're aware of? Or the lowest analog noise floor—whichever way you want to put it. Isn't it around 80?

EARGLE: No. More than that. I'm allowing the use of very artful noise reduction. I would say that normal 15-ips tape, with Dolby SR, is going to crack 100.

EDITOR: So that's 100 versus 98.1.

CLARK: Well, then there's dbx, too, which is...

EARGLE: There's dbx, which is much more audible, more of the time.

CLARK: But still, 120 dB.

EARGLE: I've never heard a peep out of SR.

(This is where we must pause. To be continued in these smaller installments.)

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Reviews of high-end electronics: Coda Technologies 01 FET preamp, Esoteric P-2 CD drive unit and D-2 multi-D/A converter, Philips LHH500 CD player, and more.

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