Issue No. 16 Spring through Fall 1991

The Audio Critic®

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In this issue:

Loudpeakers, still our favorite subject, are back in the limelight, with reviews of nine different models of widely divergent sizes and functions.

In response to the undisciplined subjectivism and lack of scientific accountability of the high-end audio press, our alternative audio philosophy is explicitly stated.

Our exposé of the wire/cable scene continues with a computer analysis of the effects of speaker cables.

We review a state-of-the-art consumer DAT deck and a collection of other sophisticated electronic components.

Plus all our usual columns and features, including choice put-downs in "Hip Boots" and a slew of CD reviews.

Issue No. 16 Spring through Fall 1991

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Cartoonist and Illustrator Tom Aczel
Business Manager Bodil Aczel

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 Critic Publications, 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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When the history of **The Audio Critic** *is reviewed at some point in the future, this may turn out to have been the most important issue. Its recipients this time include not only our current subscribers but also a much larger number of other audiophiles who are getting this one issue as a free sample. It's a promotional idea based on my perception that the main reason why a typical audiophile doesn't subscribe to* **The Audio Critic** *is that he doesn't know it exists, or if he has heard of it he hasn't ever had a copy of it in his hands. In other words, my conceit is that to see* **The Audio Critic** *is to want it. I made sure, therefore, that such a widely circulated issue defines the editorial viewpoint of the publication as clearly and comprehensively as possible. I and my small journal are what you see here, warts and all.*

* * *

Most regrettably, my plan to publish regularly at quarterly intervals in 1991 turned out to be unrealizable. The last-minute unavailability of high-quality editorial help I had been counting on was the main reason; there were also personal reasons, which at this point are no longer in force. It's quite clear that a major operational overhaul is required to make a quarterly schedule possible in 1992; the first steps in that direction have already been taken. A Winter 1991-92 issue is scheduled to come out early in the winter; the reorganization will proceed on parallel tracks. By the time the Spring 1992 issue is due, the getting-our-act-together process should be complete and the quarterly schedule automatic. That's the plan, and I have every reason to believe that this time it will work.

* * *

One of the consequences of all the delays in 1990 and 1991 is that the unpublished remainder of the "Seminar 1989" transcript is a little out-of-date, at least enough so that I'd be uncomfortable taking up a lot of pages with it. It isn't lost to posterity, however; the words remain captured and are available for some sort of future editorial use, if and when the occasion arises. Meanwhile, all you Stanley Lipshitz enthusiasts—yes, he has quite a fan club out there—can enjoy the workings of that steel-trap mind once again in the letters column starting on the opposite page. The seminar participants are still among my favorite brains for picking, and you can expect to hear from them from time to time.

Box 978

Letters to the Editor

"When men understand what each other mean, they see, for the most part, that controversy is either superfluous or hopeless," said Cardinal Newman in one of his famous sermons. This column attempts to promote understanding of what audio people really mean and thereby render their controversies superfluous or at least identify them as hopeless. 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 noticed your editorial comment about whether damaging "Letters to the Editor" get published. In July 1990, I sent you a courtesy copy of a letter I had sent to *Stereophile* in response to their inadequate and misleading coverage of the AES 8th International Conference: The Sound of Audio. Because that correspondence was completely ignored by *Stereophile,* I thought you may wish to publish it in your letters forum. *[It needed far too much additional background information to make it clear to all comers.—Ed.]*

After further reflection I wish to note that blind testing of power amplifiers has uncovered some rather important information about the response bias of audiophiles (in addition to showing that properly designed amplifiers operated within their power limits really do sound the same). In every test that used the Same/Different scoring format, listeners had a strong tendency to report Different when an amplifier was compared to itself.

Approximately 35% of the time, subjects in these tests heard differences when there were none. This is an important finding: a person with a strong interest in audio will tend to hear differences about a third of the time even when the devices being compared are level-matched and sound exactly the same. If things sound different to us even when they are the same, think about the tremendous bias toward "hearing things" when you have a coach, such as a salesman.

It's also interesting how the "wishful thinking" analysis tends to persist. Martin Colloms revisits his 1986 blind tests in the January 1991 *Stereophile.* Here he recounts how people were "shown" by statistical analysis to have been able to distinguish between two amplifiers. In fact, a cursory examination of the *Hi-Fi News & Record Review* article shows that while subjects scored 63% correct when the amplifiers were different, they also scored only 65% correct when the amplifiers were the same. About a third of the time they "heard things" that could not have been there.

Colloms based his conclusions on the correct-answer rate of the Different presentations alone. Had he included the Sames and adjusted his expected score for the response bias (i.e., subject will report Different 35% of the time even when faced with a Same), his results clearly would not support the conclusion that subjects could hear a difference.

For example, if you conducted 100 trials where amplifiers were always different, you would expect that subjects would get 35 trials correct just because they would tend to report differences even when there were none. Then, if they were just guessing, they would get approximately 32-33 of the remaining 65 right. Combined we would expect a score of 67-68 correct. Which is exactly what Colloms got.

How he imagines his 63% correct rate proves his point is beyond me. I also wonder why he never answered my letter to him raising these issues. Or why *Stereophile* didn't publish the copy of it I sent to them. Or why they didn't publish the letter I sent to them about the same subject.

I also hasten to add that we should consider very cautiously the advice given by magazines that have been unable to verify their findings under controlled conditions and resort to voodoo statistics to imply they have. If an editorial/review staff cannot fairly evaluate their own tests, what would make us think they can fairly evaluate another person's component?

Tom Nousaine
Cary, IL

You're not being singled out, Tom. When Stereophile *tried to make me look like a sleazy fly-by-night in 1988, I wrote them a letter that would have exposed their petty ill will and irresponsibility if published. The letter was highly printable in tone and very much to the point, but it nev-*

er appeared in their pages. Selective even-handedness, right?

Your criticism of the Same/Different method of blind testing is well-taken. The ABX method is far better because it has no built-in bias. The subject is asked, "Now that you've familiarized yourself ad libitum with the sound of A and the sound B, what do you think X is? Is it A or is it B?" There's no reason for anyone in that situation to lean toward either A or B.

As for Martin Colloms, he seems to be selectively scientific. When it suits him, he offers some sort of proof or technical rationale for his conclusions; at other times we just have to take his word for all kinds of off-the-wall golden-ear assertions. I think deep down he knows the truth, but he has obviously pledged irreversible allegiance to the high-end lobby, where certain truths are totally unpalatable.

And yes, I agree, 63% is a most unconvincing score. It would be unconvincing even as a bona fide bottom line, without the fudged scoring a la Colloms. When A sounds so much better than B, why can't the golden ears score 90% or even 100%?

—Ed.

The Audio Critic:

Attention all owners of Philips DAC 960 D/A converters.

We have found that approximately half of the Philips DAC960's we have encountered in the field have a serious design flaw. Specifically, the location of two critical capacitors in the de-emphasis circuitry has been swapped by the manufacturer, causing a significant frequency-response aberration (a peaking of several dB in the midtreble frequencies) in the *right* channel on all CDs recorded with pre-emphasis. This problem is easily corrected by swapping the locations of the 18 nF and 5.6 nF capacitors in the current-to-voltage conversion stage of the *right* channel.

Readers with DAC960's should ask their retailers to check their de-emphasis circuitry with an oscilloscope and a test CD, or telephone us at (515) 472-4312.

John S. Hagelin, Ph.D.
Director of Research
Enlightened Audio Designs Corp.
Fairfield, IA

A number of weeks after having been alerted to the above, Philips Consumer Electronics responded as follows:

The Audio Critic:

After contacting both our factory and our service center concerning the alleged "design flaw" of our DAC960, as reported by Mr. Hagelin, I am pleased to report that, based upon inspections of our stock, *no* mistakes were found. We will gladly repair any units found in the field that were inadvertently produced incorrectly, though we do not believe this is an issue.

Best regards,
Mike Piehl
Philips Audio Marketing Manager

The Audio Critic:

Dear Dr. Rich,

We received your letter inviting a response to the article "The Present State of CD Player Technology: Who Is Doing It Right?"

We expect to have a few "arrows directed our way" because of the fact that our basic assumptions and methods are radically different from the norm in high end digital decoding. Our research team and industry colleagues recommend that I give a brief explanation of "where we are coming from" and encourage free and open debate on the critical issues of digital decoding. Please understand that in responding to a few "arrows," we are only attempting to give an explanation for the trade-offs we have made in the design of our products—we are not attempting to defend our positions nor do we expect to "win you over," so to speak, since your positions appear to be quite polarized in many respects.

One major point of interest, as you well know, is the importance of a monotonic decoding algorithm. You state that our algorithm has "not been optimized for maximum passband flatness." (It should be noted that we have no passband ripple, which is sometimes confused with maximum flatness.) Many people argue in favor of a flat frequency response (no slight droop at 20 kHz). Fine with us, but only if and when that can be done with an algorithm that is also monotonic. (You stated the trade-off but failed to give your readers any idea as to why this trade-off was made. Even if you and Dr. Lipshitz don't personally "believe in" monotonicity, it would be only fair to explain to your readership that we do.)

Axiom #1: In the meanwhile, it is more important to great sound that the decoding algorithm be monotonic than that it have a perfectly flat frequency response.

It is so easy to kick up the response at the upper end, in order to achieve good specs. But it is not presently possible to do this and also remain monotonic. If the response is not monotonic, then you have passband ripple, echoes and, therefore, TDE (Time Displacement Error). Passband ripple would probably need to be 0.000001 dB or less not to cause time-based distortion within the digital filter, where math is often done at 36-bit resolution and then truncated.

Axiom #2: TDE is the most critical parameter separating great analog performance from the performance achieved from conventional digital decoding. (Again, even if you and Dr. Lipshitz don't personally see the importance of TDE or agree that conventional digital has a serious amount of the wrong kinds of it, a brief explanation of our position would have been helpful to your readership.)

Your readers may not know that some of our researchers have been studying the effects of TDE on audio for up to 30 years. Dr. Robert Bradford, our Chief Technical Officer, literally wrote the book on TDE as it relates to professional recording. We are enclosing a couple of the technical derivations he did for 3M-Mincom, to illustrate the fact that our team has great technical depth on this subject.

Last week, we had two Wadia VPs and a designer at my listening room auditioning a new phono cartridge we had just installed and adjusted in my system. Again, we reconfirmed the fact that we like the sound of good analog. It is involving and pleasing to listen to. We have found in repeated tests, over several years, that the only way digital decoding can compete with good analog in performance is by use of algorithms and techniques that reduce the time distortion to "about zero." TDE is the central issue in high-end digital audio.

It should be noted that we, as an engineering community, knew about this in many different ways, for many years. I first ran into TDE in 1964. At that time, M-A-K Inc. had the idea of using a Cray-1 (under development in central Wisconsin) and packet switching methods to produce a telephone digital switching system. Packet switching was abandoned, however, as a suitable technique, and later replaced with a time-accurate transmission topology by Bell Labs, due to the subjective sonic irritation of TDE. It was in the Bell Labs Blue Book where I first saw the warning that "amplitude ripple (i.e., the lack of monotonicity) causes TDE." In spite of all this, the mass-market digital people are choosing to ignore time-based issues in digital decoding. This is so they can continue to use inexpensive and simple sin *x/x* decoding methods. Wadia believes these "ripple decoders" are fine for the mass market, but we are alarmed to see such techniques masqueraded as high-end techniques. We are even more alarmed that the high-end press is allowing them to get away with it.

On the subject of "ripple decoders" we again believe your readers would have benefited from hearing both sides of the story. You state that the lack of precursor ripple on our transient response is due to the shape of our "filter," which leads to out-of-band image energy. What you didn't say was that the converse is also true. The presence of precursor ripple energy on competitive products' transient response is the result of their brick-wall filter shape. That ripple is not there for any good reason; it is a result of a filter shape. We think people should know that a brick-wall filter has about the same amount of ripple energy (you can see it clearly on a scope), during transients, right at band edge, as we have image energy above band edge. But the press only mentions image energy because it can be seen on a spectrum analyzer, whereas ripple energy averages out over time and is not seen on a spectrum analyzer. Even though it may average out over time, the instantaneous time-distortion damage is still very real. (We understand that you and Dr. Lipshitz disagree with us over the importance of precursor ripple vs. the importance of image energy. Even so, you could have stated our position so your readers understand why we made this trade-off. The reader might think the only kind of band-edge or out-of-band energy is image energy.)

What about our low-level linearity?

It is much less than one CD LSB and is, therefore, very good. It is off about 2 dB at -90 dB (5 microvolts of error, ref. 0 dBV). This has no sonic significance. (You made it sound alarmingly high! Why?)

Since you brought it up, what did we really say about the sampling theorem and the Fourier series?

First, we believe it is important for people to know that the sampling theorem clearly implies that once a musical signal is sampled (all chopped up into numbers), it is impossible to get it back "perfectly" accurate again. One reason is that the sampling intervals at the A/D and D/A must be "perfectly" the same, which is impossible (e.g., the jitter problem you and others have addressed). Another reason is that we do not have access to plus and minus infinity. The focus of our research is to get as close to perfection as possible *where it really matters.*

Second, we constantly find it necessary to remind people that Fourier never said "the world" was made up of sine waves (implied, perhaps, by the Fourier series). We simply remind them that this worldview is only an "approximation," as is clearly seen by the fact that "any theory that relies on an infinite series is, by definition, an approximation." There are *many* infinite series that can be used to model "the world." We have publicly stated that the Fourier series is not always the optimum approximation (especially in the case of inharmonic and transient musical waveforms), and we often prefer to use other approximations that are judged to be more appropriate for the design task at hand. For example, we often find it convenient to assume that "the world" is made up of an infinite number of impulses spaced infinitely close together. The Fourier series is only one of many mathematical tools, to be used as appropriate. It is not sacred!

(Dr. Rich, we feel our view is very reasonable. Your comment about this in your article appeared as though you were really "out to get us"! Why? Do you have a favorite infinite series that you are promoting today? This reminds us of some of the arguments we have had with Dr. Lipshitz. We just don't get it!)

It is interesting that you use the reference [Papoulis 1984] to refute what we say, while Abel Graham ("What's Critical in Digital," enclosed) uses Papoulis to help prove our point on TDE. Bradford suggests that this same Papoulis ("The Fourier Integral and Its Applications," 1962) was probably the first academic to warn us in high-end digital audio that passband ripple (the lack of monotonicity) leads to time distortion in transient response. I guess I had better go back and read the references again, to see whose side he is really on. (Just kidding.)

(Please understand that the purpose of this letter is to explain "where we are coming from." Taken out of context, many of the above statements would appear to be defensive or blatantly self-righteous. Please don't quote us out of context. As we see it, we are all struggling with the same issues and fighting the same technical battles. We simply approach certain problems from different vector angles, depending upon our experience and background.

So, keep up the good work, and keep the "debate" alive. As they said back in the '60s: "Let a hundred flowers blend, let a hundred schools of thought contend."

Yours truly,
Don Moses
[CEO]
Wadia Digital Corporation
River Falls, WI

Dr. Rich replies to Don Moses:

I am disappointed that you have chosen to give your standard "manufacturer's comment" reply to my article and have not addressed the points I made in the article. You continue to state that the sampling theorem is valid only for deterministic sine wave signals. As I discussed in my article, the sampling theorem is equally valid for stochastic signals, such as music. You continue to ignore the presence of a brick-wall antialias filter at the input of the analog-to-digital converter.

You state that your time-domain interpolation algorithm is superior to conventional techniques, although you have not supplied any data to justify your claims. Since the interpolators are operating in the digital domain, it should require a trivial amount of work to show that your method yields a smaller minimum mean square error (or a smaller error by any other error criterion) than the frequency-domain methods. The original objective of the work by Robert W. Moses (who is apparently not employed by Wadia), as stated in the *MONTECH* paper, was to find the optimal filter coefficients that would minimize the error in the interpolated data. According to the *MONTECH* paper, a time-domain algorithm was chosen because the optimum coefficients that would minimize interpolation error could be more easily calculated in the time domain. The paper makes no mention of TDE. Your axioms are nowhere to be found in the *MONTECH* paper. I believe that, given a sufficient number of taps, the time-domain optimization would yield coefficients very similar to a filter designed in the frequency domain. It cannot be guaranteed that the resulting optimized filter would have the monotonic property required by your first axiom, but you have sent no analytical explanation to justify the axiom.

You enclosed a number of papers by Dr. Robert S. Bradford on TDE. These papers analyze the effects of flutter and associated time-base errors on the performance characteristics of an *analog* tape recorder. This has no relevance to the interpolation and smoothing of digital signals. Dr. Bradford's work can be extended to include the effect of time-base jitter on a sampled signal. In this extended form, Dr. Bradford's work would clearly indicate the need for a small peak-jitter error in the recovered clock signal. As can be seen from my original article, I would not dispute this conclusion. Perhaps you have not fully understood that Dr. Bradford's work relates to the clock jitter problem and not the reconstruction of the digital signal. In the *preprint* of the Abel Graham newsletter (it is unclear what, if any, part of this preprint was published, and no one I know in the electronics industry has ever heard of this newsletter), TDE is discussed in the context of the interpolation and smoothing of digital signals. Mr. Graham has apparently also not fully understood Dr. Bradford's work.

The sole purpose of a reconstruction filter is to remove the image energy from the output of the DAC. If you believe that the magnitude of the image energy at the output of your decoder is unimportant, then the entire reconstruction filter can be eliminated. With the reconstruction filter eliminated, the shape of the digital impulse would show no ringing.

With regard to the droop at 20 kHz, I believe this is quite audible and will cause the Wadia player to sound less bright than competitive players. The droop could be corrected in the analog section of the player if you did not want to modify the digital filter. I note with interest that you make no offer in your letter to allow *The Audio Critic* to evaluate a Wadia product. If a unit were made available, I would perform a simple listening test. The test would be as follows: Encode an analog source with a digital recorder and compare the sound of the reconstructed output with the original source. If the digital recorder is of good quality, no difference will be heard in an ABX test. Now replace the digital recorder's DAC section with the Wadia decoder. I believe that the Wadia will be clearly audible in the ABX test. This test would conclusively show that the Wadia decoder is changing the sound of the original source.

With regard to low-level linearity, I indicated in my article that gain linearity provides only a limited amount of information on the DACs performance. Harmonic distortion measurements are much more important. A Wadia X-32 was found to have 30% harmonic distortion at -90 dB by *Stereophile* (Aug. '90, Vol. 13, No. 8, p. 125). Competitive state-of-the-art products are now using DACs with almost unmeasurable harmonic distortion at -90 dB (PS Audio, Theta, Meridian, and Harman/Kardon, for example). I am amazed that you find it acceptable to produce a $7995 decoder box which has poorer low-level harmonic distortion performance than a $200 CD player with MASH DACs. The effects of low-level linearity errors are the only significant measurable differences between modern CD players. I am surprised that you find a passband ripple of 115 nV unacceptable (the increase in magnitude of a 1 V rms signal subjected to a 0.000001 dB level increase) yet find a 5 µV gain-linearity error (relative to 1 V rms) reasonable. Since Wadia uses high-quality DACs, it is quite possible that the low-level linearity problem is the result of a software "bug." That would be good news for Wadia owners because, once the bug is identified, the problem could be fixed by changing an EPROM.

Finally, I want to point out that Dr. Stanley Lipshitz reviewed my manuscript; he was not a coauthor. Any similar comments made to you by Dr. Lipshitz regarding your product's design are independent of my analysis.

David Rich
Contributing Tech. Ed.

Dr. Lipshitz replies to the Editor:

I would like to respond to Don Moses' letter for three reasons: (a) Although I am not the author of the article on which he is commenting, he repeatedly addresses his remarks to both David Rich and myself ("you and Dr. Lipshitz"); (b) I agree with most of the statements made by David Rich in his article, and in particular with those concerning Wadia's decoding algorithm, to which Don Moses takes exception; and (c) I have on numerous occasions expressed to Don Moses my belief that his company is fundamentally misguided in its digital filter design.

The first point to note is that, in the analog reconstruction process, one is attempting to recover as accurately as possible the original analog signal whose samples have been recorded. Of the infinity of such analog signals (yes, there are indeed infinitely many signals which have the *same* samples; they are all aliased versions of one another), there is only *one* which is bandlimited to the Nyquist frequency (one half of the sampling frequency). This unique analog signal is the one which we should be trying to reconstruct. It is the bandlimited signal whose samples were taken in the original analog-to-digital conversion process. (The input antialiasing filter did the required initial bandlimiting.) These statements are the essence of the sampling theorem. The earliest proof of the sampling theorem of which I am aware was given by E. T. Whittaker [1] in 1915. Whittaker presents a very general and profound result which includes the proofs of the statements made above. The uniquely represented analog signal is what he calls the "cardinal function." Now, the essential point is that the cardinal function is obtained from its samples by a sin *x/x* reconstruction process, this being the time-domain equivalent of a brick-wall bandlimiting filter set at the Nyquist frequency in the frequency domain. (The Fourier transform of a perfect brick-wall filter is a sin *x/x* function.) The process of removing the "images" of the Nyquist band by means of this brick-wall filter results in the reconstruction of the originally sampled analog signal. This is a mathematical theorem.

I must thus reject Wadia's claim that there is something improper or deficient or inappropriate in trying to approximate as closely as feasible a true sin *x/x* reconstruction. This ideal reconstruction filter (which can only be approximated) must pass without change all frequency components up to the Nyquist frequency (i.e., it must have a flat passband with linear phase response up to the Nyquist frequency) and completely remove all frequency components (the "images") above this frequency. It must thus approximate to a brick-wall filter, and the extent to which it fails to do this is a measure of the error it makes in the reconstruction. Note, by the way, that it does not matter whether the brick-wall filter is all analog (as in early digital audio systems), or partly digital and partly analog (as in current systems). It must be there.

This brings me to my second point. Given the above, why does Wadia use a reconstruction filter which significantly attenuates the high audio frequencies (by 3 dB) and passes a goodly chunk of the out-of-band images? Moreover, why do they maintain that their reconstruction is more accurate than a sin *x/x* reconstruction? (More accurate to what?) I believe that it is a misguided approach, based on an approximation to the wrong criterion. A cynic might be inclined to speculate that the treble cut masquerading as greater accuracy is the audible reason why some people might "prefer" this less accurate (to the original samples) sound. But inaccurate it is. It does not come close in any sense to approximating the ideal brick-wall filter discussed above. Most digital audio reconstruction filters are much closer. So what lies behind Wadia's filter design? It is an attempt to make the filter's impulse response more compact in time than the ideal sin *x/x* filter's oscillatory time-domain behavior. This attempt seems to be based on the belief that there is something inherently wrong with the latter, but as the sampling theorem shows, this is not the case. The pre- and postringing of the ideal brick-wall filter does *not* in *any* way introduce precursors or postcursors (?) which were not already present in the original bandlimited signal which is being reconstructed from its samples. To believe otherwise is a serious misunderstanding of the mathematics involved, and hence of the true outcome. This may seem counterintuitive, but it is correct. For example, if the input analog signal was bandlimited by a *causal* brick-wall filter approximation (e.g., a minimum-phase analog antialiasing filter), which thus had no precursors in its impulse response, an ideal sin x/x reconstruction will *not* introduce any precursors .

It seems that Wadia believes that the ringy nature of a sin *x/x* impulse response is inherently undesirable (whereas, as I argue above, it is actually correct), and so sets out to design a filter with less pre- and post-ringing, which is what they have done. But you cannot have it both ways. To the extent that your filter departs from a sin *x/x* impulse response it is reconstructing a modified (read "wrong") version of the signal. In their pursuit of a filter time-domain response closer to the mistaken goal of a perfect nonbandlimited impulse with no overshoot or ringing, they have attenuated the top half of the audio band and also allowed substantial ultrasonic garbage out (which is nonlinearly related to the original analog signal). If Don Moses really believes that a single-sample-high impulse response is ideal, he can very easily achieve it. Just omit the reconstruction filter entirely, and allow the baseband and all images out unattenuated! Why does Wadia not do this? Because then, of course, you don't get an analog-looking signal back—you get back the sampled waveform with all its discontinuities, a far cry from the original.

To summarize, I maintain that Wadia's approach to digital-to-analog conversion is inherently flawed because of what appears to be a misunderstanding of the sampling theorem itself. Don Moses' desire for a monotonic frequency response would seem to be simply a reflection of an unnecessary constraint, which forces his system into the errors that it makes as a result of the Lagrangian interpolation used. In no way do I accept Don Moses' two "axioms." Maybe Wadia ought to reassess them. (Note that an axiom is not a provable result, but an *assumption* from which results can be deduced.) Finally, the sampling theorem relies, not on the Fourier series as claimed by Don Moses, but rather on the Fourier integral, which *does* imply "an infinite number of impulses spaced infinitely close together." There is no contradiction inherent in the use of the sampling theorem.

Yours sincerely,
Stanley P. Lipshitz
Audio Research Group
Departments of Applied Mathematics and Physics
University of Waterloo
Waterloo, Ontario
Canada

Reference:

[1] E. T. Whittaker, "On the Functions which are represented by the Expansions of the Interpolation-Theory," *Proc. Roy. Soc. Edinburgh,* vol. 35, pp. 181-194 (1914-1915).

As Editor, I want to make absolutely sure that the reader understands the essential thrust of the professorially restrained commentary by the two academics above. To put it less politely but more simply than they do, Wadia Digital is designing and selling D/A conversion equipment based on incorrect mathematics. The Moses versus Rich/Lipshitz debate is not about some kind of legitimate diversity of informed opinion but about mathematically provable fact. (By the way, that bit about "a hundred flowers"—that's Mao in the '50s, not "they" in the '60s.) Assuming that Moses is presenting the rationale of his technical team accurately—and there's always the possibility that he isn't—Rich and Lipshitz are clearly right, Moses is clearly wrong, and Wadia DIA conversion is clearly faulty. It's as simple as that—as long as Moses puts it as simply as he does—and no authoritative scientific opinion exists to the contrary.

—Ed.

The Audio Critic:

David [Rich],

...[Regarding] your article: The main point of contention I have is with your comment that "the jitter level of the YM3623 is sufficiently low...." It is hardly sufficient. The jitter problems I alluded to earlier were indeed the result of using the YM3623 in the manufacturer's recommended circuit. There are tricks one can play, however, to adjust the circuit around the YM3623 to improve its performance, but the part, by itself, is basically junk. Unfortunately, it is also the only low-cost commercially available S/PDIF interface chip out there, so I guess we have to live with it until Yamaha, Crystal, and Philips all get their PLLs to work. (The new Yamaha receiver is over a year behind schedule because of this problem, and the Crystal part is overdue as well, I suspect for the same reason.)

Secondly, your statement that a "brick-wall" filter has a sin *x/x* response is patently false. The impulse response of any system is, of course, the frequency response of the system; for the filter, it would be the filter's response characteristic. The impulse response of the *sampling system* is sin *x/x,* in consequence of the finite sampling time. So let's put the blame where it is due, and leave my poor analog filters alone!

Power supply bypassing, rather than separate regulation, is all that's necessary for good performance on our DACs. The problem we've found most people have is that they use poor bypassing and layout techniques—and often end up using regulators as a crutch to solving their problems. This is an extra expense that could have been saved if they took the time to lay out their circuit properly in the first place.

Other than these minor points, your article was comprehensive as well as well written. I'm even thinking about subscribing to *The Audio Critic,* after having seen this issue. It seems to be the only "audiophile" magazine I've seen that takes a pragmatic approach to audio....

With best regards,
Rick Downs
New Product Development Engineer
Audio Products
Burr-Brown Corporation
Tucson, AZ

Dr. Rich replies:

The brick-wall filter I referred to was the *ideal* brick-wall reconstruction filter required by the sampling theorem. An ideal brick-wall filter has constant magnitude in the passband below the cutoff frequency f_c and total rejection above the cutoff frequency f_c. A lowpass filter of this form (which is not realizable) will have a sin *x/x* time-domain impulse response. The sin *x/x* time-domain impulse response in the sampling theorem is a result of this ideal reconstruction filter. As Mr. Downs indicates, a realizable brick-wall filter will not have a sin *x/x* response. The sin *x/x* response is closely approximated by the digital filters used in modern CD players.

The sin *x/x* frequency response which Mr. Downs refers to is a property of a real DAC that I did not discuss in my original article. In the ideal sampling theorem, the sampled signals are assumed to have the form of an impulse function. In practice, each sample at the output of the DAC has a finite width (Mr. Downs refers to this as the finite sampling time) and is approximately rectangular in shape. This characteristic of a real DAC can be modeled as an ideal DAC followed by a fictitious filter. The frequency response of this filter will have a sin x/x response. This additional filter response results in a small high-frequency rolloff. This drop is compensated by the digital or analog filter in a CD player.

David Rich
Contributing Tech. Ed.

The Audio Critic:

...You saved me a lot of money, as I was about to purchase a Meridian 208 CD player/preamp.

I learned a great deal from Dr. Rich's essay but was disappointed that the CD reviews said that it makes no difference what is inside the players—they all sound alike. Did some players not show any difference in soundstage or image focus? What about depth??? Yes, you saved me a lot of money. But now I don't know what the hell to buy. Thanks.

Ralph Riutti
Moorpark, CA

Let's be precise. I never said it makes no difference what's inside the players; on the contrary, I discussed at some length the measurable differences in electronic performance, as well as the differences in construction quality and ergonomics. What I did say was that, within the group of 13 units reviewed in that particular issue (No. 15), I and my associates found no audible differences in the course of a somewhat limited number of ABX comparison tests. On a previous occasion (see Issue No. 12, p. 36), we heard, and I reported, a tiny difference in one instance.

As for soundstage, image, depth, etc., you must understand—as the audio pundits whose golden ears are attached to muddled heads do not—that those "structural" qualities of reproduced sound are determined by the recording site, the microphones, the microphoning and mixing techniques used, the signal processing added (if any), and the radiation characteristics of the playback loudspeakers—not by the design of the playback electronics. You can safely assume that a reviewer who waxes eloquent over the soundstaging or depth of an electronic circuit has no serious credentials as a technical expert. Even the easily measurable difference between 15-bit and (almost) 16-bit resolution in digital audio appears to be audible only on special test signals and not on music.

—Ed.

The Audio Critic:

I would like to add a few personal comments to the discussion between yourself and William J. Roberts [Issue No. 15, p. 7] on the subject of constant-directivity speakers.

There is no connection between how something is recorded and how it is reproduced. I have never seen a report of someone changing from a pair of bipolar speakers, used to listen to music recorded with ribbon microphones, to a pair of omnidirectional speakers to listen to music recorded with omnidirectional microphones, or changing to a pair of cardioid-pattern speakers (there are such things) to listen to music recorded with cardioid microphones.

Likewise, there is a similar lack of reports of stacking speakers on top of each other to listen to music recorded with coincident microphones, or moving them apart to listen to music recorded with spaced microphones.

There is no connection between how music is monitored during recording and how it should be reproduced.

In the production process, there are such techniques as LEDE™ and RFZ™ in use. These are two of the several methods used in trying to listen only to direct arrivals when analyzing sound. One attempts to absorb the room reflections in foam material, and the other attempts to steer the direct and reflected sound so that reflected sound arrives too late to be perceived. To these may be added the use of the famed 604 monitor speakers and the use of headphones, both of which provide a preponderance of direct sound and little or no reflected sound.

The reason for this is that reflected sound results in what may be called spaciousness, low interaural cross-correlation, or diffusivity, which are all related concepts and which correlate with listener preferences.

In my opinion, it is difficult for most people to find fault with something which gives them pleasure, which is generally a useful trait but does interfere with the business of monitoring recorded sound, using a reproduction system with characteristics which listeners find pleasant.

Mr. Roberts mentioned the research results of Floyd Toole. One of Dr. Toole's findings was that the preference of his listeners correlated positively with measured increases in beamwidth and with measured constancy of beamwidth with frequency. He did not claim to test anything that was called "constant directivity," but neither did he mention any complaints of "brightness" going along with increases in beamwidth and beamwidth constancy.

Yes, a measurement microphone will give a higher reading at higher frequencies when measuring the output of speakers which do not get too beamy at high frequencies, since it picks up both direct and reverberant fields. Simplistically speaking, human hearing responds to direct sound for amplitude information and to reflected sound for spatial information. So, an increased high-frequency reverberant field should not make an amplitude difference.

There have been a few consumer loudspeakers which were actually constant-directivity speakers, although not identified as such. They were favorably reviewed but did not seem to remain in production for long. An example is the Genesis 44, which seems to me to have been quickly replaced with a more profitable "improved" model, to which one could add examples such as the AR MGC-1 and the original dbx Soundfield, which also had a cardioid-like pattern. "Brightness" was not a complaint.

Speakers with constancy of dispersion angle do have a problem, though, related to perceived localization.

Much classical music is recorded with spaced microphones. This results in a recording which sounds just fine when played back on beamy speakers. The localization is adequate to identify a source location for the string section, and there is plenty of what seems to be " hall sound."

But, when played on nonbeamy speakers, the sound field takes on an unbelievable shape, or rather lack of shape. For instance, a violin solo is "right over there," and also other places. There is no "edge" to the sonic image, so that it sort of "blurs away" towards the sides. In other words, the phase differences that result in pleasantly low interaural cross-correlation from beamy speakers result in a loss of localization information from nonbeamy speakers. This is simply unacceptable to some people.

There is a solution: learn to like music recorded with "original instruments," which were less loud, used in smaller spaces to produce the same volume as later instruments, and generally recorded from a single point, for practical reasons. Then, speakers that are nonbeamy are not bothersome. In fact, I rather like the ones I have.

Regards,
James P. DeClercq
Roseville, MI

You oversimplify. Yes, the playback geometry in standard practice is totally unrelated to the recording geometry—but no, that's not necessarily a desirable situation, nor is it invariably the case. For example, the original Edison phonograph was a system of sound reproduction in which the *recording geometry and the playback geometry were perfect mirror images of each other—and that was the best part of an otherwise highly limited system. The same kind of symmetry exists today in binaural recording and playback: the headphones replace the microphones in exactly the same position, without any change in geometry. Another symmetrical technique is to take a small group of, say, 8 performers and close-mike each of them with 8 separate microphones feeding 8 separate channels and tracks, then play the 8-track tape back through 8 speakers deployed in the same relative positions as the original performers. It isn't practical and it's rarely done, but it can be very lifelike indeed.*

As for monitoring during recording, the frequency response of the monitor speakers obviously affects the decision of the producer as regards the correct frequency balance. If the monitor speakers have a rolled-off top end, an inherently overbright recording will sound just fine in the control room but not in the home through flatter speakers. That much of a "connection" between monitoring and playback at home is self-evident; as for the relative brightness of constant-directivity speakers, I'll admit that in a large, well-padded room the issue may be moot, but in a small or medium-sized room with hard surfaces the increase in reflected high-frequency energy will not be sufficiently separated from the direct sound to avoid the impression of increased brightness—I have experienced this myself

I'm inclined to agree with you, on the other hand, on the subject of spaced microphones and the trade-offs they entail. I can live with those trade-offs, however, especially since single-point microphoning has its own characteristic shortcomings.

—Ed.

The Audio Critic:

...I was delighted with the contents of Issue No. 15. The article by David Rich on CD player technology is, in my opinion, the finest article on the design of a CD player I have read and one of the best on audio technology I have ever seen. One of the major virtues of this article is that it is not condescending nor oversimplified nor obtusely technical. I read it as I would a good mystery—on the edge of my seat. What I have learned from it makes me feel one up on the trash that appears elsewhere in the audio press and the pseudosophistication of some high-end sales people....

I have been following the double-blind test desert storm for some time. The people who oppose it or question its validity remind me of the pharmaceutical-firm vice presidents who fought this same approach for testing new drugs—they are either dumb and/or know that they have a lot to lose from the objectivity that is forced on them by double-blind studies. The basic emotion common to audio gurus and pharmaceutical manufacturers in this context is greed. Without double-blind studies a lot more dangerous or useless drugs would be in circulation. Too bad that there is no FDA for audio. I write that tongue in cheek, as it takes ten years and $10 to 100 million to market a new drug. Perhaps an *Audio Critic* would suffice for us lovers of music.

Cordially yours,
Steven E. Mayer, Ph.D.
Nashville, TN

I really don't believe that the "audio gurus" you refer to are motivated primarily by greed. There are better arenas for greed than high-end audio. (Try a used-car lot or a massage parlor.) No, you're talking about self-indulgent, posturing little people looking for groupie approval and protecting the belief system of the cult. They're more worried about their ego than about their money—although vested interest can't be entirely ignored—and they aren't big enough to admit they were wrong even when the facts are incontrovertibly demonstrated to them. On the contrary, the more the scientific audio community snickers at their voodoo, the more they try to proselytize those who know even less than they do. You say they're greedy and manipulative; I say they're untutored, fuzzy-minded, insecure, and unaccountable.

About David Rich's article, I agree with you 100%. He is a great addition to our staff. And that's just the beginning; other highly accredited people will soon be coming on board.

—Ed.

Who Sued Whom and Why: *Stereophile* and Carver Corporation in Court

I keep getting all sorts of inquiries about last year's mysterious lawsuit between *Stereophile* and the Carver Corporation. Most of the inquirers are under the impression that Carver sued *Stereophile.* Not so. Maybe *Stereophile* would like the audio community to believe that, but that's not what happened. No audio manufacturer in his right mind (except, of course, Bose) would want to be the plaintiff against the free press in a suit about a bad review. The actual fact is that *Stereophile* sued Carver.

Why did they sue? Basically because they didn't understand what they were getting themselves into. Carver had run a 12-page ad in the May/June 1990 issue of *The Absolute Sound,* in which six Carver amplifier reviews were reprinted. One of them was Robert Harley's hatchet job on the Carver "Silver Seven-t" in the January 1990 *Stereophile;* the five others were highly favorable reviews, including one of mine. The ad made *Stereophile* look kind of stupid—like the schoolyard bully. Thereupon *Stereophile* sued Carver Corporation for copyright infringement, claiming the latter had no right to reprint the review without permission. Carver responded the standard way, by filing a countersuit.

Stereophile's suit was essentially frivolous and fell on its face in court—the text of an attack is not protected by copyright when reprinted in the context of a defense against it. Carver's countersuit, on the other hand, had some legal substance and threatened to bankrupt *Stereophile* if pursued to the finish. The claim was that a systematic pattern of maliciously discriminatory "Carver bashing" and recklessly irresponsible/incompetent equipment reviewing had caused multimillion-dollar losses to Carver Corporation. It could have turned into a First Amendment battle, but the court ordered the parties to go into arbitration first.

The outcome was a rather astonishing settlement, "with prejudice." For three years—1991, 1992, and 1993—*Stereophile* is forbidden to put the word "Carver" in print, just about regardless of context, with minor legalistic exceptions. They basically have to pretend that Carver doesn't exist. Carver, in turn, is forbidden during those three years to discuss publicly the alleged deficiencies of *Stereophile's* equipment testing or to disseminate reprints of the disputed *Stereophile* reviews, again with some additional minor legalisms.

The gist, as I interpret it: *Stereophile* signed away its First Amendment rights to prevent a possible disaster in court and in exchange received what was most important to it—silence on the touchy subject of its competence. A class act, eh? *—Ed.*

A Loudspeaker Miscellany: Big Boxes, Satellites, Dipoles, Subwoofers

By Peter Aczel
Editor and Publisher

While there are no ultrahigh-end models in this group (the most you can spend here is two thousand and change), the best of these units raise serious doubts about the need for insanely expensive speakers.

Those who are familiar with my loudspeaker reviews know that I like to make some deep philosophical remarks and broad generalizations before I get down to the nuts and bolts of specific models. Now, I can't keep repeating myself at the beginning of each new loudspeaker survey (like this one) just to communicate to new arrivals where I'm coming from philosophically, so I must refer first-time readers to earlier issues, especially Nos. 10, 11, and 14. I do want to bring up here, however, a point I haven't made perfectly clear (if you'll pardon the tainted expression) before.

Where are the curves?

When it comes to speaker systems, I don't particularly like to show frequency response curves and other graphic displays of performance. A frequency response curve is fine and dandy for showing, say, the de-emphasis error in a CD player or the characteristics of a parametric equalizer, but it can be quite misleading in the case of a loudspeaker system. A difference of a few inches in microphone placement can make a tremendous difference in the measured response curve—very flat and smooth this way, quite jagged that way—and the audiophile looking at the curve in a magazine will jump to erroneous conclusions either way. Rigid measurement protocols—such as aiming the microphone at the geometric center of the speaker system, or at the tweeter, or at the woofer, from a distance of one meter, three meters, etc.—will result in superficial and inconclusive data. Each speaker system tends to be a law unto itself and must be measured with a certain flexibility of technique that comes from experience. Formularized measurement with pat graphic output as its goal is poor audio journalism, at least in my opinion. (When loudspeakers become as predictable as amplifiers, I'll change that opinion.)

My method is to use the B&K microphone more or less as a doctor would use his stethoscope, poking and probing every which way, near and far, at the "sweet spot" and at the not-so-sweet spots, using all sorts of test signals and monitoring everything on the spectrum analyzer and/or the oscilloscope. Pretty soon I have a very good idea of just how smooth the response is, whether there are trouble spots (ringing, lobes, phase reversals, etc.), how deep the bass goes, whether the output from the various drivers coalesces into a semblance of coherence and at what point, and so forth. No, it's not as perfect a technique as I would like—and, yes, I do take fixed on-axis and off-axis measurements at set distances, but I don't entirely trust them. A large and systematic family of curves taken in an anechoic chamber, which I don't have, would probably be preferable but still subject to audiophile misinterpretation if published; the gated pseudoanechoic measurements favored by some reviewers also have serious limitations. I contend that my eclectic method arrives at the qualitative truth—and isn't that the reviewer's truth?—without fail and with a high degree of objectivity, even if it leaves something to be desired quantitatively from the engineering researcher's point of view.

Oh, yes, in all fairness, there is a routine, formularized measurement which is very accurate, namely the Don Keele method of extreme-nearfield bass response measurement. It tracks the anechoic curve beautifully up to 100 Hz or so, and I bow toward Elkhart, Indiana, every time I avail myself of Don's great little shortcut. Wouldn't it be nice if it worked equally well at higher frequencies?

Cambridge SoundWorks Model Eleven

Cambridge SoundWorks, Inc., 154 California Street, Newton, MA 02158. Model Eleven portable satellite/subwoofer/amplifier stereo system, $749.00. Tested sample on loan from manufacturer.

Does this music system in a small suitcase belong in a survey of loudspeakers? Well, where else does it belong? Its salient qualities all have to do with the speaker designer's art; electronically it's rather conventional. Designed by Henry Kloss (a legend in his own time—or in his own mind, depending on your perspective), the Model Eleven represents some clever engineering in the size-versus-performance department. The so-called BassCase, a piece of hard-sided luggage just under 20" long, holds all the pieces —6½" high satellite speakers, 7" wide amplifier, all sorts of cables and adapters, optional Walkman or Discman—and when emptied becomes the airtight enclosure for the built-in 7" acoustic suspension woofer. Truly a virtuoso shoehorn job. One little problem is that casual, sloppy repacking just doesn't work; only the fastidious neatnik will be able to close the repacked case. (There goes the youth market.)

I listened to the Model Eleven at some length, in the company of several associates, before I measured it, and we came to the conclusion that for an ultracompact trick system it sounded remarkably complete and accurate but not quite as good as the best conventional stereo systems of only slightly larger size and comparable cost. For travel by automobile, these other systems will fit into the trunk with equal ease though perhaps not as neatly; for travel by plane, bus, or train the Model Eleven is of course unbeatable. One thing I faulted in the performance was an unpleasant shattering on piano music; the little amplifier appeared to be clipping on the peaks. On less dynamic music—cool jazz, for example—it's really a classy-sounding little system.

I must confess that I was unable to take this equipment as seriously as, say, a Snell speaker system, so that my measurements were not very extensive. I did determine that the BassCase woofer goes down to 42 Hz before starting to roll off—very respectable for luggage. The electronic crossover network incorporated in the amplifier is specced to operate at a crossover frequency of 150 Hz; from the response of the satellites it looked more like 200 Hz to me, but even 150 Hz is a little high for completely nondirectional L+R bass. The 3" midbass/midrange driver in the tiny satellites appeared to be extremely flat in its range; the ¾" dome tweeter, on the other hand, registered a very rough response through the perforated metal grille (nonremovable and possibly the sole cause of roughness). The crossover to the tweeter is in the neighborhood of 4 kHz, as far as I could tell by poking around in the nearfield. The amplifier can also be powered from the 12-volt DC cigarette lighter socket of a car, but my aversion to elaborate auto sound kept me from trying it that way. (I believe that one should be listening to engine and road sounds with at least one ear when driving and not be ecstatically plugged into Wagner or Jerry Lee Lewis, oblivious to the audible world outside.) A 9-volt DC power takeoff for your Walkman or Discman is on the back of the amplifier.

As you can probably see from the above, I'm not the right customer for the Cambridge SoundWorks Model Eleven, but the right customers do exist, and I think they'll be very pleased with the system. Henry Kloss still knows how to juggle and massage the size/performance/price trade-off.

Carver "Amazing Loudspeaker" Platinum Mark IV

Carver Corporation, P.O. Box 1237, Lynnwood, WA 98046. "The Amazing Loudspeaker" Platinum Mark TV, $2199.00 the pair. Tested samples on loan from manufacturer.

This unique loudspeaker has become something of an obsession for Bob Carver, as indicated by the fact that the Platinum Mark IV is its fifth-generation version (not counting the experimental versions that never went into production). On the back cover of Issue No. 15, a review of the Mark III was announced as one of the coming attractions, but Bob has meanwhile fiddled with the crossover network and the frequency balance once again, so we're now looking at Mark IV. He says this is "It" now, no more changes, but I'm skeptical. Not about the basic design, though.

I've said it before and I'll say it again: this is a classic, a landmark design that rewrites the book in a number of respects. It's the first open-baffle loudspeaker system without active equalization to come even close to state-of-the-art bass performance. It's the first loudspeaker system to use a monolithic ribbon-type line-source transducer successfully all the way down to 100 Hz. (By successfully I mean without serious irregularities in response.) It's also the first genuinely clean *large-signal* loudspeaker system at anywhere near its price. In other words, it's a breakthrough—in deed, not just in claims. In a large listening room its dangerously addictive qualities really assert themselves; I find even the very best conventional, forward-firing enclosed speakers somehow downsized and uninvolving by comparison, and I soon go back to the Amazing. I'll never look at speakers in the five-figure bracket the same way again because many of them are simply not as good as this $2199 system.

Since this is my third review of Bob's brainchild—see Issue No. 11 for my evaluation of the original version and Issue No. 14 for the Platinum Mark II—I really don't want to go over the same ground once again. New readers are advised to obtain those back issues for a more thorough discussion of the underlying design principles. Here I just want to make a few additional comments and note the latest changes.

I occasionally hear off-the-wall theoretical objections to the Carver open-baffle bass system, which is the speaker's most ingenious feature. You just have to listen to it, but some people don't seem to trust what they hear or they don't understand the concept. It's almost too simple to be plausible. An open baffle necessarily creates a 6-dB-per-octave low-frequency decline. A woofer with an abnormally high Q will have a big bass bump. The bump can be tailored have a 6-dB-per-octave rise. Eureka! The two opposite slopes will cancel out to create a flat response. In practice, of course, it's not so simple. The shape and dimensions of the open baffle, the woofer Q and resonant frequency, the voice coil and cone must all be precisely designed and held to tight tolerances, or the whole schmear just won't track as a system. In fact, that's just what happens in planar speakers with inerently high-Q bass panels—Magneplanar, Apogee, etc.—which manage to produce a bass of sorts thanks to the same laws of nature, but such a bass is not nearly as flat and correctly damped as in the Carver because that "eureka" perception of mirror-image slopes is not part of their design.

One of the pseudotechnical objection I've heard is that, well, it's still a high-Q woofer, and we all know that means underdamped. Wrong. The open baffle acts as an acoustical short circuit, which lowers the Q analogously to an electrical short circuit. Or, you could say that instead of combining a conventional low-Q woofer with a conventional high-Q box to produce the desired system Q, the Carver accomplishes the same thing by combining a high-Q woofer with a low-Q (and how!) open baffle. The difference is that the response profile corresponding to the desired Q in the Carver emerges only after the acoustical cancellation has taken place, a small distance in front of the speaker; the extreme nearfield measurement still shows the high-Q bump (obviously, the Don Keele method isn't applicable to open-baffle systems). Now, the Platinum version of the Amazing has four 12" woofers per side, a total of eight, and the fundamental resonance after break-in is in the neighborhood of 22 Hz, at which frequency the response is still essentially flat. That combination of air-moving capability and low-frequency extension results in absolutely majestic, life-size bass reproduction. No owner of the Amazing will ever need to bring up the subject of subwoofers. In Mark III and Mark IV, the acoustically derived equivalent Q is continuously adjustable on the rear panel from 0.5 to 1.0 (no more resistors to insert, as in Mark II). Another change in Mark IV is that the woofers and the ribbon all move forward in response to a positive-going pulse, thus satisfying one of my well-known little compulsions.

About those rear-panel controls—there are three of them now and they work very nicely, but I disagree with the way they are marked. The leftmost one is the Q control, and its "recommended" start-up position is marked with a calibration line at 1.0 (all the way up clockwise). The flattest bass response I measured in my large listening room was obtained with the control at 0.7 (12 o'clock), confirming the theoretical prediction. The 1.0 setting sounded too heavy. The middle control has a range of approximately 6 dB for adjusting the upper midrange, and its calibration mark is at 9 o'clock, whereas the measured flattest setting in my room was at 3 o'clock. The far right control, with a similar range, trims the high frequencies, and the calibration mark is again all the way up clockwise. I had to back it off slightly to about 4 o'clock for flattest response. I other words, the start-up "recommendation" favors a shallow U-shaped frequency response, with heavy bass, crispy highs, and a recessive midrange—the kind of balance I generally associate with unsophisticated hi-fi jockeys, who at the same time want to be told that "everything is flat." Well, it's flat my way, not their way. And here's what I mean by flat: I had to use my Audio Control third-octave real-time spectrum analyzer instead of my trusty old Hewlett-Packard sweep spectrum analyzer because a 5-second log sweep is useless in the farfield in a live room—and, as I indicated, this particular speaker must be measured in the farfield—whereas pink noise analyzed in real time still gives a more or less reasonable reading. Between the 1-dB-per-step and 2-dB scales I estimated that the response was ±1.5 dB from 25 Hz to 20 kHz—that's the range of the instrument—at about 4 meters. Not too shabby! Bob Carver claims that outdoors, where a much more precise reading is obtainable, he can find a sweet spot where the speaker is so flat, with the controls trimmed in, that nobody would believe him if he published the curve—but I believe him after my own measurements. Of course, only the very best program material sounds best when reproduced dead flat, but that's an old problem—and its solution isn't a nonflat speaker.

Alvin Foster of the Boston Audio Society, with whose perceptions about loudspeakers I nearly always agree, wrote a 12½-page evaluation of the Amazing in *The BAS Speaker* (Vol. 18, No. 1), covering various aspects of the subject in much greater depth than I could ever hope to with my review work load. I recommend this massive article—less rigorous than an AES paper but more so than an underground audiophile review—to all interested parties. (Address: The BAS Speaker, P.O. Box 211, Boston, MA 02126-0002.) Alvin confirms my previous findings as regards the large-signal capability, extended bass response, uncommonly low distortion, and tremendous clarity of the speaker, but concludes that those are not the main reasons for its superior sound. What then? He claims it's the dispersion or polar pattern and "incredibly" flat overall frequency response. That seems to contradict, at first blush, a recent mathematical analysis of line sources by Stanley Lipshitz, who is not in the habit of being wrong. Alvin has entered into a dialogue with Dr. Lipshitz to try to find out why the Carver ribbon has mysteriously better response than the mathematical model would seem to permit. I'm sure there's a nonmystical explanation. I know, for example, that the ribbon is passively equalized within the crossover/control network to compensate for certain inherent acoustical radiation effects, but there may be more to it than just that.

One minor annoyance I found in successive incarnations of the Platinum version is a tendency to develop very high-Q breakup resonances near the two ends of the ribbon. These buzzes are heard only when the speaker is swept with sine waves at a fairly high level; on music there's no problem. Mark III was already very much cleaner in this respect than Mark II, and in Mark IV the fault appears to have been cured entirely. That ribbon is basically nothing more than Reynolds Wrap glued to a plastic membrane, crinkled, and stretched on a frame between magnets; it takes some production experience to keep it 100% stable.

I also want to emphasize again that (1) the Carver speaker must be pulled well into the room, at least 3½ to 4 feet away from the back wall, to produce the kind of sound

I've been talking about and (2) it must be well broken in—meaning about 50 hours of dynamic music—before it will give you its absolute best performance. Your local dealer probably won't have satisfied those conditions when you ask to hear the speaker, and he probably won't be driving it with over 500 clean watts per side as I do. I have no control over his particular situation or yours; I'm just telling you what happens in my listening room.

Let me conclude with something I may not have fully communicated before. Consider the most exalted ultrahigh-end loudspeaker systems in the world: the Infinity IRS and IRS Beta, the top-of-the-line Martin-Logans, the Sound Lab A series, the Wilson Audio WAMM, the Thiel CS5, the Duntech Sovereign, the Apogee Diva, the new B&W Matrix 800, and others in that general bracket. Don't for a moment imagine that any of them is strikingly or overwhelmingly better than the Carver "Amazing Loudspeaker" Platinum Mark IV, that stepping up to one of them from the Carver is to step into another world. No way. I'm not about to make comparisons here; you may end up preferring this one or that one for various reasons—or even the Carver over all of them, for the reasons already discussed. The point is that the Carver is right up there—almost as good, better by a hair, not really as good, or what have you, but in there—at $2199 the pair! That constitutes a very serious political problem which the high-end community isn't ready to deal with. By solving certain long-standing technical problems in a dramatically cost-effective way, Bob Carver has created as much of a monster for the industry as a boon for audiophiles. History will end up being on his side, but for the moment the high-end community is not. I've always enjoyed the spectacle of second-rate minds freaking out over a first-rate reality, so I'm having fun with the Carver monkey wrench in the high-end mystique, but sooner or later that reality will have to be accepted by all rational audio people. Don't wait until then, however, to check out the Carver speaker and form your own opinion. Life is short.

JBL XPL160A

JBL Consumer Products, Inc., a Harman International Company, 240 Crossways Park West, Woodbury, Long Island, NY 11797. XPLI60A floor-standing 3-way loudspeaker system, $2498.00 the pair. Tested samples on loan from manufacturer.

A major paradox of the loudspeaker industry: JBL makes the best drivers, has the slickest production techniques, and is both progressive and honest in the R & D area—yet there seems to be no truly first-class JBL speaker system for home use (as distinct from professional sound). The XPL series is supposed to be JBL's breakthrough in the audiophile market, but on the basis of the XPL 160A I can't confirm that. It's a frustrating, self-contradictory speaker.

In my review of the JBL L40t3 two issues ago, I called the proprietary pure-titanium 1" dome tweeter in that system the best known to me, bar none. I have to reiterate that opinion now, after having tested an updated version of the same tweeter in the XPL160A. And that's not all. The midrange driver in the XPL160A is designed around a pure-titanium 3" dome, a tour de force never before attempted to my knowledge, certainly not as successfully as in this remarkable unit. How they got rid of all the standing waves is beyond me, but they did. The two dome drivers are mounted as close together as possible and crossed over at approximately 4 kHz to form what functions, in effect, as a single seamless 1 kHz to 20 kH transducer of the utmost flatness and smoothness. The two voice coils are wired out of phase, probably as a concomitant of a second-order network. There's no high-frequency peak; the rolloff begins at 20 kHz but stops and reverses a bit after 30 kHz. The older version of the tweeter went out a few more kHz on axis but wasn't quite as well damped, and the off-axis rolloff began sooner. The double-dome combination has just about the same response 30° off axis as on axis, meaning almost perfectly flat (when the microphone axis is at the most favorable height) and showing much smaller squiggles—maybe 2 dB from peak to peak—than I've seen in other drivers. No trace of ringing of any kind, either. If I were in charge of a new project to design a conventional electrodynamic speaker system, these are the drivers I'd like to specify because they're simply the best; unfortunately JBL keeps them strictly in-house.

So far so good—indeed, fantastic. The 10" woofer in its vented box is another matter. The 33" high cabinet itself is gorgeous—high-gloss black lacquer finish, subtly nonparallel side walls (trapezoidal cross section), neoprene-lined baffle step to time-delay the domes (very impressive craftsmanship), elaborate open grille frame, and so forth—but the tuning of the woofer enclosure appears to be less than optimal. The box frequency (where the displacement of the woofer cone is at a minimum) is 34 Hz, but maximum output from the rearward-facing vent is at 44 Hz, and that looks like the effective low-frequency cutoff of the system. I've seen deeper bass out of smaller boxes at the same efficiency (between 88 and 89 dB SPL at 1 meter with 1 watt input).

The paper cone of the woofer is, as far as I can tell, the downfall of this speaker system. Its frequency response is extremely flat, but there are—you guessed it—energy storage problems. Tone burst tests revealed definite ringing in the octave just above the crossover frequency of 800 Hz, where the 12 dB per octave rolloff begins. (The 3" dome actually comes in just above 1 kHz, also with a 12 dB per octave slope, but for some reason there's no hole in the summed response, perhaps because the woofer and midrange are wired in phase despite the second-order crossover.) If the otherwise excellent woofer were used only up to, say, 400 Hz, there would be no problem, but with the 800 Hz crossover the ringing in the 1 kHz to 1.5 kHz range is insufficiently attenuated and becomes the signature of the speaker. I have a feeling that a you-must-use-what-we-have corporate policy was imposed on the engineers here. For all

I know, the woofer was conceived for a totally different application; it's very good, for example, in terms of linear excursion—the Q doesn't change at all as the amplitude of a step-function input is increased.

The sound that results from this mixed bag of design elements is intriguing but ultimately unsatisfactory. Above 2 kHz or so everything is utterly neutral, transparent, and smooth as silk, as good as you'll ever get out of a forward-firing system. The barely attenuated ringing immediately above the woofer's passband, however, is a pervasive coloration at all times and on all types of music. It comes off as a breathy hollowness, and it's right there in the midrange where you can't get away from it. I find it to be a disqualifying fault of what would otherwise be a stupendous-sounding loudspeaker, albeit somewhat light on the bottom end. The top-of-the-line XPL200, which has been promised to me for review, has a 12" woofer and a separate lower-midrange driver covering the two octaves from 300 Hz to 1.1 kHz, so it probably avoids the same pitfall. Those titanium-dome drivers deserve the best possible system design.

Snell Type C/IV

Snell Acoustics, Inc., 143 Essex Street, Haverhill, MA 01832. Type C/IV floor-standing 3-way loudspeaker system, $2190.00 the pair. Tested samples on loan from manufacturer.

In Issue No. 13, I wrote that "the Snell Type C/II is just about a state-of-the-art 'monkey coffin' (trade slang for a conventional forward-firing one-piece speaker system in a rectangular box)," and in Issue No. 14 I followed that up with a fairly detailed review explaining why I think so. Type C/IV is a successor model in exactly the same format, so I won't start at square one here to describe the system; you may want to refer back to the C/II review. (In case you wonder about Type C/III, it was discontinued almost as soon as it was announced.)

So now the Snell Type C/IV is the state-of-the-art monkey coffin. Yes, it's better than the C/II in a number of ways. The bass is greatly improved; I measured a classic fourth-order Butterworth response—vent response peak precisely filling in the woofer null at a box frequency of 24 or 25 Hz. That's quite an achievement with a 10" woofer, an internal volume of less than 3½ cubic feet (estimated), and fairly high efficiency (88.5 dB). I'd say Kevin Voecks is "pushing the envelope," as the saying goes in high-tech country. He wasn't when he did the bottom end of the C/II. The large-signal step response of the C/IV woofer is a bit more Q-ey than the small-signal step response, indicating less than perfect linearity on long excursions, but nothing in this world is perfect. It's still a very impressive 10" bass system.

The front tweeter of the C/IV is also new and ostensibly improved, although the old soft-dome unit was certainly good enough. The Vifa metal-dome tweeter now favored by Snell is claimed to be the best representative of the breed; I find that JBL's proprietary titanium dome is even better, but the Vifa is indeed extremely flat and smooth in response. It begins to roll off at 18 kHz, then kicks up again and comes to a high-Q peak at 25 kHz. That's fairly typical of metal domes and completely unobjectionable to me. (I don't know how my dogs feel about it because I keep them out of the laboratory.) Front tweeter level is continuously variable; the calibrated Optimal position of the level control appears to be accurate. The rearward-firing little Audax supertweeter (for "air and balance") remains unchanged, as is the on/off switch for it. And, yes, the two pairs of terminals for bi-wiring are still there, in genuflection to unscientific cultism by an otherwise scientific company, but—what the hell—they do no harm.

My measurements clearly indicated that the frequency response of the total system is optimized/normalized to the axis of the midrange driver, where the deviation from absolute flatness is no more than ±2.5 dB, maybe only ±2 dB. That's for the full range from deepest bottom to tip-top—truly remarkable. Off-axis response remains almost as flat over an impressively wide angle. (See also David Rich's review of the Snell Type Q in this issue for his observations about Snell's design approach, their QC procedures, and their use of Floyd Toole's NRC facilities in Canada.) Tone bursts swept through a wide range of frequencies revealed negligible storage. Woofer, midrange, and tweeter are connected in phase—a positive-going pulse makes them all move forward—but a square pulse input cannot be acoustically recovered from the speaker regardless of where the microphone is placed. That, of course, is the nature of the beast—a 3-way system with 4th-order Linkwitz-Riley crossovers—and Snell has never considered pulse coherence to be a design requirement. (This is not the time and the place for a dissertation on the audibility or inaudibility of phase.)

The sound of the Snell Type C/IV is, yes, the best I've ever heard out of a monkey coffin—uncolored, transparent, low in distortion, high in resolution, perfectly balanced, much better on the bottom end than the C/II. The frequency balance—which depends not so much on whether the response is ±2 dB, or ±2.5 dB, or whatever, but on just where those little zigs and zags occur—is probably the most satisfying of any speaker known to me. To be sure, there's more to a speaker than frequency balance—for example, the Carver "Amazing" at exactly the same price produces a larger, more authoritative, more dynamic, more concert-hall-like sound—but even the Carver could use the C/IV as a model in the frequency balance department.

To say something negative—I have to search for it—the cabinet quality could be a little higher considering the price, not so much in basic construction but in little details of finish. I have a feeling that the C/IV is somewhat costlier to make than the C/II, and something had to give. Don't let that stop you from giving very serious consideration to this outstanding loudspeaker system.

Velodyne ULD-15 Series II

Velodyne Acoustics, Inc., 1746 Junction Avenue, San Jose, CA 95112. ULD-15 Series II subwoofer system (single unit) with power servo controller, $1795.00. Optional passive highpass/bypass accessory available. Tested samples on loan from manufacturer.

This subwoofer system has been around for quite a few years, but I never had a chance to get my hands on one. More recently its designer formalized his ideas on the technical aspects of the subject in an Audio Engineering Society paper (David S. Hall, "Design Considerations for an Accelerometer-Based Dynamic Loudspeaker Motional Feedback System," 87th Convention of the AES, New York, NY, 18-21 October 1989, Preprint 2863), which rekindled my interest and motivated me to start bugging the company for a review sample until I got one. What they sent me was the Series II update of the original 15-inch model (12-inch and 18-inch versions also exist).

My curiosity about the Velodyne must now be balanced against my well-known—or at least frequently avowed—reluctance to go over the same ground that someone else has already covered with great competence. I'm referring to David L. Clark's exhaustive and authoritative six-page review of the original ULD-15 in the November 1987 issue of *Audio,* free reprints of which are available from Velodyne. There's really very little I could say about the almost identical Series II that DLC hasn't explained in considerable detail. The difference appears to be mainly a beefed-up version of the external power servo controller, which now has a rated amplifier power output of 400 watts rms and incorporates some circuit changes. Between the Hall paper and the Clark review, there's no opening left for piercing insights by your Editor. Even so, I must attempt an appreciation (in the literary sense), a full-fledged technical test report being clearly unnecessary.

The system is probably the most highly refined embodiment of the motional-feedback approach to bass transducer design, the principal advantage of which is greatly reduced harmonic and intermodulation distortion. The Hall paper claims an improvement by a factor of 10 over conventional woofers, and the Clark review confirms that. It should be noted, however, that conventional woofer distortion is largely excursion-related, so that the Carver "Amazing Loudspeaker," for example, with its four 12" woofers per side achieves comparably low distortion figures simply by dividing up the total excursion requirement among a larger number of feedbackless drivers. The uniqueness of the Velodyne is that it allows almost any reasonably good pair of speakers to acquire ultralow-distortion bass, flat all the way down to the limits of hearing, and takes up only 2½ square feet of additional floor space. The ULD-15 is normally delivered with the active lowpass and highpass filters in the controller unit set to 12 dB per octave slopes and a nominal crossover frequency of 85 Hz. Other frequencies, as well as 6 dB per octave slopes, are available as a dealer-installed option. A further option is the passive highpass/bypass switching box, which allows the main speakers to be crossed over passively with 6 dB per octave slopes or to be played full range without the subwoofer. Thus there exists more than the usual degree of flexibility in the main-to-sub marriage, although David Rich's caveats on that subject (see the article that follows) still apply.

I was particularly interested in how the ULD-15 would complement the Quad ESL-63. That's one great speaker that can definitely use bass extension to live up to its full potential. The default frequency of 85 Hz for the crossover point seemed about right, as it overlaps the bottom end of the ESL-63 by an octave, eliminating the need for sophisticated matching and allowing confident use of the active 12 dB per octave lowpass and highpass sections. The audible results were excellent, even with just one ULD-15, although I would have preferred two. Directionality is not an issue below 85 Hz, but room excitation at two points, 3 dB down each, will produce a less aggressive complex of standing waves than single-point excitation at full power; furthermore, two subwoofers provide 3 dB more headroom on bass transients and generally tend to give a more complete impression of the low-frequency characteristics of the concert hall. Velodyne, however, is promoting the monolithic matrixed subwoofer concept, so reviewers get one unit and that's that. At any rate, the transition between a pair of Quads and a single Velodyne appeared to be quite seamless to me—yes, tweaks, the ULD-15 is "fast" enough for the electrostatics, whatever that means. (There's no such thing as a fast woofer, boys and girls. If a woofer were fast, it would be a tweeter. I think semieducated audiophiles mean a well-damped woofer without hangover when they use the word. Motional feedback certainly achieves that.) Despite the smoothly and deeply extended bottom end, the Quads still don't sound like big speakers with lots of headroom. They sound like Quads with deep, clean, detailed bass. That's far from the worst thing that can happen to a music lover, to be sure, but a large-signal Quad is not yet a reality.

Level matching to the main speakers is a *sine qua non* with the Velodyne, although at the CES and in other commercial demonstrations the level is always cranked up to show off the amazing bass, so everything sounds thick and unnatural. With the 85 Hz crossover and 12 dB per octave slopes, proper level matching means that you'll hear no difference at all on certain kinds of music when the subwoofer is bypassed and the main speakers are allowed to play full range. Only when there's real bass should you hear any, but then you should hear it life-size and perfectly defined. To obtain that kind of correct level adjustment in a real-world listening room, your ears are probably the best instrument, especially if you move around the room and experiment with many different recordings. Remember that bass is the foundation of music but not its sole purpose.

Of course, you're probably aware of the audiophiles whose taste is so exquisite that they regard all bass as vulgar and unnecessary. The Velodyne is definitely not for them. 0

In Your Ear


WHO VACUUMED UNDER MY CABLES?
JOE! TELEPHONE, DEAR.
YOU KNOW I CAN'T COME TO THE PHONE WHEN I'M TESTING!
KEEP THAT DOG OUT OF HERE WHEN I HAVE EQUIPMENT LYING AROUND!
BILLY, DON'T YOU DARE TOUCH ANYTHING!
THESE ARE NOT YOUR TOYS!
MOMMY.... WHY IS DADDY SO ANGRY?
OH, DON'T MIND HIM, HONEY. HE'S JUST RELAXING WITH HIS STEREO.
TOM ACZEL

The Minimonitor Reexamined: Four Current Examples

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

As a format, the small but accurate loudspeaker is alive and well and living with us in the digital era. Here a longtime user of minimonitors evaluates four noteworthy designs, from $550 to $1000 the pair.

Editor's Note: *My almost 4000-cubic-foot listening room has made me somewhat impatient with small box speakers; they somehow never seem to fill the airspace to my satisfaction, with or without subwoofers. To give these widely respected minimonitors a fair hearing, I decided to ask David Rich to review them, even though his primary discipline is solid-state electronics rather than electroacoustics. He is heavily into minimonitors, however; they suit his listening environment, and unlike so many loudspeaker reviewers he knows the laws of physics and is undaunted by filter theory. I did the measurements on the speakers, partly by myself, partly in his presence. Where I differ with him on a particular point, or just wanted to put in my two cents worth, I have inserted editorial comments in brackets. The usual alphabetical order of brand names has been dispensed with to preserve David's train of thought.*

* * *

Introduction

Since the introduction of the Rogers LS3/5A in the 1970s, the minimonitor has become an important option for the audiophile who wants good sound at a reasonable price in a small-sized package. The principal advantage of the minimonitor is the small baffle on which the drivers are mounted. The small baffle reduces secondary radiation of sound from the speaker's surface. Owing to the monitor's small size, cabinets with few resonances can be designed. The result is a speaker which is more difficult to localize and offers a more focused presentation of instruments within the soundstage. The speaker's small size and light weight also give the listener greater flexibility in positioning the speaker in the listening room. When not in use (or when company comes), the speaker can be moved to a less conspicuous position.

The principal disadvantage of a minimonitor is limited low-frequency extension. The physics of a loudspeaker system will prevent any speaker with a radiating element of 6-inch diameter and a 72-cubic-foot box from producing appreciable bass below 100 Hz. Early minimonitors attempted to extend the low-frequency response by using an underdamped bass characteristic (see *The Audio Critic,* Volume 1, Numbers 4 and 5 under the old nomenclature). This extended the low-frequency limit but resulted in a frequency-response peak of as much as 6 dB in the midbass. Another approach to extending low-frequency response is to reduce the efficiency of the system. This approach requires a larger displacement of the woofer cone at a given specific pressure level. In turn, there is a higher level of distortion when reproducing bass and midbass signals. Another source of increased distortion is the wide frequency range required to be reproduced by the minimonitor's woofer. This range is typically from 75 Hz up to 3 kHz, spanning five octaves. This results in increased intermodulation distortion in comparison with a 3- or 4-way speaker, where the midrange driver is usually required to cover a range of only two or three octaves. The higher distortion levels that occur in a minimonitor at a given pressure level—relative to a larger speaker system—limit the ability of the minimonitor to achieve realistic reproduction at loud signal levels.

The low-frequency limit of a minimonitor is clearly audible in most types of music. It is not just the double bass, organ, and tuba that are affected. Male voice, cello, trombone, bassoon, and piano are all significantly lightened in timbre. The low-frequency limit of a minimonitor can be extended by the use of a subwoofer. A limited number of minimonitors are designed to be used with dedicated subwoofers. Most minimonitors do not offer this option; instead, the problem of combining the subwoofer and the minimonitor is borne by the user. Combining the two speakers is not a trivial problem:

- A crossover frequency must be chosen.
- The levels of the subwoofer and minimonitors must be matched.
- The transfer characteristics of the highpass and lowpass sections must be determined.
- The transfer characteristics of the speakers in the crossover region cannot be ignored [Leach 1980].

To minimize the impact of the speakers' transfer characteristics, it is advantageous to cross over in a range where the speakers are flat in frequency response for at least an octave above/below the crossover frequency. To prevent localization of the subwoofer in single-subwoofer systems, it is necessary to cross over to the subwoofer below 100 Hz. This, unfortunately, requires the crossover to take place in the transition-frequency region of the minimonitor. A professional speaker designer would use optimization software and sophisticated test equipment to design the crossover network. These tools are not available to end users or most dealers. As a result, the performance of many "roll your own" minimonitor/subwoofer systems is mediocre.

One approach to crossing over the speakers adopts a high-order (e.g., fourth-order Linkwitz-Riley) network [Linkwitz 1976]. Interference effects between the subwoofer and the minimonitor are then minimal. It is difficult to synthesize a below- 100-Hz high-order crossover network because the values of the required crossover components become uneconomical. An active crossover and an additional amplifier are required to cross over a speaker at these frequencies [Bullock 1985]. To avoid the effect of the transfer characteristic of a minimonitor with a 60 to 75 Hz cutoff (-3 dB) on the frequency response of the composite system, the crossover should be placed at 130 to 150 Hz. This frequency, however, is too high to prevent localization of the mono subwoofer. A lower crossover frequency can be used if the complex poles in the minimonitor's transfer response are canceled by a pair of complex zeros synthesized in the active crossover network [Linkwitz 1980]. The required position of the zeros is minimonitor-specific; hence, the active crossover must be customized for each minimonitor.

Another method of crossing over the minimonitor at a lower frequency uses the transfer response of the minimonitor as part of the crossover. If a sealed-box minimonitor, with a second-order transfer response and a Q of 1, is combined with a first-order highpass filter that has its -3 dB frequency set at the natural frequency of the minimonitor, then a third-order Butterworth highpass filter characteristic will be created. The subwoofer is rolled off with a complementary third-order Butterworth lowpass section. The resulting summed output will be flat in frequency. The summed transfer response will have a second-order allpass characteristic; thus, the resultant system is not minimum phase. The sensitivity of the ear to phase variation remains a point of controversy [Lipshitz 1982], [Fincham 1985], [Deer 1985], [Greenfield 1990]. Assuming the nonminimum-phase characteristic is not very audible, the principal disadvantage of this approach is the merely 6-dB-per-octave electrical rolloff of the minimonitor woofer. As a result, harmonic and intermodulation distortion caused by the presence of low-frequency signals is not markedly reduced in this approach. In addition, since the subwoofer and minimonitors are physically separated by relatively large distances, the drivers can constructively and destructively interfere with each other. Care must be exercised in positioning the minimonitors relative to the subwoofer. Hiding the subwoofer behind a piece of furniture located in the corner of the room is not a prudent decision. In my experience, the subwoofer must be placed equidistant between the minimonitors for a seamlessly integrated sound. For this review of minimonitors, I used this approach to cross over to my subwoofer. The subwoofer was the Spica Servo (discontinued). In this subwoofer the third-order lowpass filter is preset to 65 Hz. This is because the subwoofer was designed to cross over to the Spica TC-50 minimonitors. The other minimonitors had a resonant frequency close enough to the TC-50 that I was able to get a good blend for all units under test. The Velodyne subwoofer reviewed in this issue also uses a passive highpass section as one of the available options and an active lowpass section.

Spica TC-50

Spica, 3425 Bryn Mawr N.E., Albuquerque, NM 87107. TC-50 compact 2-way loudspeaker system, $550.00 the pair. Tested samples owned by reviewer.

I have used the Spica TC-50 as my reference for seven years. The Servo woofer was added six years ago. Before the availability of the Servo, I used the Janis W-2 subwoofer and the Janis Interface 1A crossover. (The Janis system was reviewed in Volume 1, Number 2—old nomenclature—of *The Audio Critic.)* While the Janis was capable of astonishingly low bass, it did not blend well with the Spica, and the woofer was easy to localize. The Servo woofer blends perfectly with the TC-50 and is nearly impossible to localize. It does not have the low bass of the Janis, however. Two years ago I replaced the original set of TC-50's with a newer version which is representative of the current production. The TC-50 is a remarkable value at $550.00 the pair. The level of sound quality achieved by Spica with the low-cost Audax paper-cone woofer and soft-dome tweeter is admirable. The speaker reproduces pulses as well as any multiple-driver dynamic speaker we have tested. A tone burst test revealed no major resonances in the drivers.

The Spica achieves good pulse response by attempting to minimize the change in phase angle of the reproduced signals over the frequencies in the speaker's passband. So far, no conclusive research has shown that the ear is very sensitive to allpass filters (filters that change phase only and have constant magnitude). Studies—see the last four references cited above—have indicated that the ear is partly sensitive to phase distortion at low and midrange frequencies.

First-order crossovers are capable of summing to a flat magnitude without introducing an excess phase response, i.e., the system is in minimum phase [Lipshitz 1983]. For this potential benefit of a first-order crossover, you gain a pair of real disadvantages. These disadvantages are a consequence of the shallow slope of the transition band, only 6 dB per octave. This means that the drivers receive significant energy in frequency bands that they are not required to reproduce. As a result, intermodulation distortion products increase and driver resonances may be excited. The second disadvantage is that the drivers will acoustically add or subtract in the crossover region, as a consequence of their differing positions on the speaker baffle. This results in a narrow vertical angle over which the speaker will have flat frequency response. In a first-order crossover this effect will occur over a two-octave range because the drivers are rolled off slowly. The effect is magnified by the fact that in a first-order crossover (and other odd-order Butterworth crossover networks) the first null occurs just 15° above the axis of the loudspeaker (at a crossover frequency of 3000 Hz, with drivers whose centers are separated by 4.5 inches), and the maximum radiation level (the main lobe) occurs 15° below the speaker axis [Bullock 1985]. Small changes in vertical angle will cause large frequency variations in the crossover region. In addition, John Bau of Spica argues that the downward-tilted main lobe significantly increases the amplitude of signals reflected from the floor in the crossover region.

In practice, it is difficult to create a first-order crossover in a two-way system because the natural rolloff of the drivers occurs near the crossover center frequency. The actual slope of the transition band of the drivers will be closer to 12 dB per octave, or even 18 dB per octave. Spica circumvents these problems by using a fourth-order Bessel lowpass crossover on the woofer. The Bessel filter introduces an approximately constant time delay in the passband of the woofer. The tweeter is then physically displaced behind the woofer to compensate for the woofer's time delay. The natural rolloff of the woofer is included as part of the fourth-order network. The phase shift from the tweeter's highpass crossover section (first-order in the Spica) and the natural rolloff of the tweeter must be accounted for if the total system is to exhibit the desired minimum phase characteristic. By optimizing key parameters, Spica achieved the goal of flat on-axis frequency response and near minimum phase characteristics over a band from 350 Hz to 4 kHz. Ralph Gonzalez has independently analyzed the Spica approach [Gonzalez 1988] and verified that the bandwidth requirements of the drivers are less stringent than in the case of a first-order crossover. Gonzalez reports that a lowpass Bessel filter only approximates a pure time delay; the Spica approach "cannot quite match the phase integrity of an ideal first-order crossover." Because Spica incorporates the natural rolloff of the drivers into the crossover, it is necessary to individually match a woofer, a tweeter, and a set of crossover components to achieve the desired results. This is an amazing procedure for a speaker that retails for $275 per side. Factory records are kept for each speaker. Replacing a driver is simply a matter of Spica sending a replacement driver of similar characteristics. The Spica approach does not eliminate the problem of a narrow vertical angle over which the speakers have a flat magnitude response. This problem exists because the drivers still overlap over a relatively wide frequency range. Our measurements confirm Spica's recommendation that the speaker stand be set so that the center of the woofer is at ear level.

With the speaker positioned at optimum height, the Spica presents a wide soundstage with good localization of instruments. The midrange has little coloration when the speaker is used within its sound pressure limits. Instruments are reproduced with good air and ambience, and with minimal box coloration. On the downside, the speaker is not as detailed as the Audio Concepts Sapphire II (see below). The upper midrange is especially affected, sounding slightly closed-down and blurred. The top end was not as clean and open as that of the Audio Concepts, although the sound of the Spica was less bright. The upper midrange and treble colorations are most noticeable on close multimiked recordings. These recording methods represent the vast majority of classical recordings available today. The colorations from the Spica made these recordings sound more edgy and fatiguing to listen to than on the Audio Concepts, despite the Audio Concepts' brighter presentation. The frequency response of the Spica has a slight downward tilt. This may explain the loss in detail. The tilted response allows the speaker to sound balanced even when used without the subwoofer. In conclusion, the TC-50 is an excellent value in the $550 price class. Many speakers in this price class have better bass response, including Spica's own SC-30, but none that I have heard have an equally uncolored and boxless presentation of the midrange.

Audio Concepts Sapphire II

Audio Concepts, Inc., 901 South 4th Street, La Crosse, WI 54601. Sapphire II compact 2-way loudspeaker system, $789.00 the pair (direct from Audio Concepts, fully assembled, including shipping charges). Tested samples on loan from manufacturer.

The Audio Concepts Sapphire II must be ordered directly from the manufacturer. By bypassing the dealer, the speaker can be purchased at lower cost. This is not equivalent to purchasing a speaker at cost, since the cost of customer service, normally assumed by the dealer, is now assumed by the manufacturer. Audio Concepts gives a 30-day in-home trial period. This is adequate to determine the quality of the speakers in your room. What you will not be able to determine is whether other competing models would serve your needs better. This is a function that the high-end dealer serves by offering several competing models in a given price range and advice on which model will yield the best sound quality for your listening room and musical tastes.

There is no question but the Sapphire II is a great value. *[It's pretty much the same speaker as the Sapphire reviewed in Issue No. 14, minus the elaborate and unwieldy system of 2" acoustical foam and with a redesigned cabinet, permitting considerable savings.—Ed.]* The cabinet is ¾" medium-density fiberboard with extensive bracing. Using the nonscientific knuckle test, it appears to be nonresonant. Crossover components are exclusively polypropylene capacitors and, with one exception, air-core inductors. Expensive Focal drivers with Kevlar cones are used. The speaker has separate banana plugs for both the woofer and tweeter. This allows bi-wiring the speaker to the amplifier. There is no physical reason why bi-wiring should improve the speaker's sound. The speaker is shipped with a pair of shorting wires between the woofer and tweeter input terminals to allow the use of a single speaker cable. The four input terminals are arranged in the form of a perfect 0.75" square. A double banana plug can be placed across the shorting wire by plugging it into two red or two black terminals by mistake. The result is a direct short across the amplifier. The distance between the input terminals connected by the shorting wire should have been increased, so that this occurrence would not be possible.

The Sapphire II, like the Spica TC-50, reproduces pulses with outstanding fidelity. Tone burst tests again revealed no major resonances in the drivers. THD measurements were also very similar to those of the Spica. If the Focal drivers have any advantages over the Audax drivers used in the Spica, they were not apparent in our tests. *[As a matter of fact, the Focal tweeter has a 17 kHz resonance, which the Audax does not, but the Focal has better off-axis response.—Ed.]* The Audio Concepts Sapphire II achieves good pulse response by using first-order crossover sections. The manufacturer of the speaker recommends a lower position at the bottom of the woofer as the optimum vertical axis. *[Confirmed.—Ed.]* Jack Caldwell, the designer of the Sapphire II, used optimization techniques to overcome the problems in traditional first-order crossover design. The computer, however, does not replace the requirement for a skilled designer. Many optimized networks will exhibit unacceptable sensitivities to changes in driver parameters. Often a program will converge to local minima on the performance surface, not the global minimum. The results of the optimization are also only as good as the parameters entered into the program. According to Audio Concepts, both on-axis and off-axis data for the drivers were used in the simulation. Emphasis was placed on achieving flat magnitude and phase response on a horizontal axis of 15° to 30°. *[The 30° off-axis response is superb.—Ed.]*

In the Sapphire II, a second pole is added to the tweeter network, a decade below the crossover frequency, to prevent distorting the tweeter with large levels of low-frequency energy. In the woofer network, additional reactive components counteract the natural rolloff of the woofer slightly above the crossover frequency [Gonzalez 1987]. If this rolloff were not taken into account, a frequency response error in the crossover region would result. The problem with this network is that the woofer is driven at full power above 6 kHz. This is in contrast to the Spica, which rolls off the woofer at a 24 dB per octave rate. The Sapphire II sounds very unpleasant when auditioned on axis. This effect may be partially attributable to the fact that the woofer is receiving full power at high frequencies. We observed some frequency response aberrations in the tweeter on axis that disappeared off axis. This may also be responsible for the subjective on-axis problems. *[The woofer is acoustically active up to about 4 kHz on axis. The off-axis improvement in the tweeter response consists mainly of the attenuation of the 17 kHz peak.—Ed.]* Center fill can often be improved by toeing in a pair of speakers. Because of the response problems on axis, you cannot toe in the Sapphire II's.

The enclosure of the Sapphire II uses aperiodic loading. This is a fancy name for a ported system with a heavily damped port. The result is a transfer response which closely approximates that of a sealed system. The claimed advantage is reduced cone movement at system resonance. Audio Concepts adds another complexity to the bass system by using a woofer with dual voice coils. The second voice coil is driven only at low frequencies by a separate crossover network. This second voice coil compensates for a diffraction loss that results from the small baffle of the minimonitor [D'Appolito 1988]. Modern low-frequency speaker design theory (developed by Thiele and Small) assumes a hemispherical space. This can be approximated by a baffle with large dimensions compared to the longest wavelength radiated. This is not the case for a minimonitor placed away from the back and side walls. With a minimonitor operating at low frequencies, some energy diffracts to the rear, around the baffle. The specific pressure level of the speaker falls as the effective volume into which the speaker radiates increases. The second voice coil is used to increase the driver output in order to correct for the diffraction loss. I have not seen any AES papers that advocate aperiodic loading, although I can report that the Sapphire II has a more detailed midbass response in comparison with the Spica. It is unclear whether the subjective improvement is from the aperiodic loading, the diffraction correction, or just the low resonance of the Audio Concepts cabinet.

The Spica, which uses a simple acoustic suspension bass loading, has a low-frequency rolloff 10 Hz lower (at 65 Hz) than the rolloff of the Audio Concepts. The Spica also has a system Q of 0.9. The Audio Concepts has an equivalent second-order farfield Q of 0.5, according to Jack Caldwell. The lower Q is claimed to offer better-damped bass. In practice, this low Q, in combination with the 75 Hz rolloff and the slightly hot top end, creates a speaker which has a sound too light in weight to be listened to in the absence of a subwoofer. The low Q creates problems interfacing this speaker with a subwoofer that uses a passive highpass section. As stated above, this approach assumes a speaker with a Q of approximately 1.0. A high-order electronic crossover would allow a good match to a subwoofer. The use of such a crossover would result in significant excess phase in the crossover region. This defeats one of the principal design goals of the speaker, but the effect is probably not very audible [Fincham 1985]. Another approach is to assume that the rolloff of the speaker is approximately first-order because of the low Q. A passive first-order filter can then be cascaded with the speaker to form an approximation to a second-order Linkwitz-Riley highpass section. The woofer is crossed over with a complementary second-order Linkwitz-Riley lowpass section [Estrick 1991].

Audio Concepts is just introducing a subwoofer (Sub 1) which has been optimized to work with the Sapphire II. This subwoofer must be used in pairs. Its passive crossover network uses a second-order electrical lowpass section in combination with the acoustical characteristics of a close-to-the-floor downward-firing driver position to achieve what the manufacturer calls a "synthesized bandpass" acoustical transfer function. The synthesized acoustical bandpass filter is claimed to have a flat passband response from 90 Hz down to approximately 20 Hz, without the size and cost penalities of more conventional design approaches. A passive first-order highpass section rolls off the satellites at 100 Hz. We plan to test the Sub 1 in a future issue of ***The Audio Critic.***

In summary, this is an excellent speaker when used in conjunction with a good subwoofer. I found the Sapphire II to offer sufficient improvement over the Spica to justify the extra $239 expense, provided the brighter balance of the speaker is acceptable. After eight years, I decided to sell the Spicas and purchase the Audio Concepts. The comparison of the Spica with the Audio Concepts is not quite fair, since the retail price of the Audio Concepts, if it were sold at retail, would be approximately $1300. John Bau, Spica's chief designer, is currently at work on a new upscale minimonitor which has drivers, crossover components, and a cabinet equivalent to, or possibly of better quality than, those of the Sapphire II. The final battle of the war between the minimum-phase minimonitors has not yet begun.

* * *

After this set of reviews was completed, Audio Concepts announced a change to the Sapphire II crossover, which lowers the tweeter level by 3 dB, as calculated from the changed component values. *[Mike Dzurko of Audio Concepts told me it was only about 1.5 dB; I never had a chance to measure the modified units, but David Rich asked for and wired the mods into his review pair.—Ed.]* This change results in a more natural presentation of the vast majority of recordings in my collection. Wonderful recordings, such as the Rachmaninoff symphonies conducted by Eugene Ormandy (CBS Masterworks) and the Shostakovich string quartets performed by the Borodin Quartet (Angel), are now very listenable. This more than compensates for the small loss of air in the treble on some audiophile recordings. *[The tweeter level looked about right to me on the spectrum analyzer before the mod, and I was happy with the way it sounded. David apparently wants his speakers to act as a tone control on overbright recordings. Unfortunately, that's a one-size-fits-all tone control.—Ed.]* The upper midrange is less affected by a tweeter level change than in the Snell Type Q (see below) because the first-order crossover of the Sapphire II creates a broad transition between the woofer and the tweeter. As a result, the response change more closely resembles a tilt rather than a shelf. Production units manufactured after March 1991 incorporate the change.

Snell Type Q

Snell Acoustics, Inc., 143 Essex Street, Haverhill, MA 01832. Type Q compact 2-way loudspeaker system, $780.00 the pair. Tested samples on loan from manufacturer.

The design philosophy Snell Acoustics applied to the Type Q minimonitor differs radically from Audio Concepts' and Spica's. High-order crossovers (fourth-order Linkwitz-Riley) prevent driver interference in the crossover range and reduce intermodulation distortion [Linkwitz 1976]. The speaker is optimized for the flattest possible frequency response. Each crossover is individually optimized to ensure that each speaker matches the prototype response profile within ±0.5 dB. This optimization process takes into account the unit-to-unit variations of the drivers used in each speaker. The speaker was designed at the NRC research facility in Canada, one of the premier loudspeaker testing facilities in the world. The design clearly benefited from the fund of knowledge established through the extensive research work of Dr. Floyd Toole and his staff at the NRC, although they were never directly involved in any Snell project. Double-blind testing methods are used in evaluating the subjective performance of Snell speakers. Based on these tests, Snell has concluded that the excess phase that results from high-order crossover networks is not significantly audible on musical program material. Snell also concluded that a conventional sealed box with a Q of 0.7 would yield good subjective bass characteristics in this model. Despite all the science, the speakers have tweaky bi-wiring provisions.

Snell uses high-quality drivers in the Type Q. A polypropylene-cone woofer is combined with a modified Vifa tweeter. Cabinet construction is of high quality, but the box is lighter than that of the Sapphire II and does not appear to be as extensively cross-braced. The front baffle of the Snell is shaped so that the grille frame is flush with the baffle. This, along with the rounded grille frame, eliminates sharp edges that could result in diffraction. This is an advantage over the conventional grille used by Audio Concepts. Spica uses a large felt pad on the front baffle, which fits into the grille frame to control edge diffraction.

A noteworthy feature of the the Linkwitz-Riley crossover is the vertical radiation pattern, which is symmetrical about the speaker's vertical axis. A speaker that uses the Linkwitz-Riley crossover has a spectral balance which is relatively invariant with the listener's ear height relative to the center of the speaker. This clearly works to the Snell Q's advantage, since the sound of the speaker changes minimally as its height above the floor is varied.

Given the pedigree of the system, I was ready to be awed and overwhelmed by its sound. Unfortunately, this was not the case. The speaker sounded edgy because of excess treble energy. The boxes would not disappear as the Spica and Audio Concepts did. The sound was not as open and lacked the air presented by the other speakers. A tweeter level control is provided on the back of the Type Q. This control allows the overall level of the tweeter to be raised and lowered. This shelf equalization, however, does not solve the speaker's problems. Turning the tweeter control down to a point where the speaker was naturally balanced with respect to the top end resulted in an upper midrange depression that removed much of the life from instruments and vocalists. What is required is not a shelf equalizer, but a method to slowly tilt down the upper midrange and treble of the speaker.

[My measurements showed a very slight upward trend in response—rising without peaks maybe 1 dB per octave above 3 kHz—with the tweeter control in the recommended 12 o'clock position, and a gradually developing broad dip centering on 6 kHz or so as the control was turned down, without significant effect on the upper highs. Overall, though, the response was flatter and smoother than that of the Audio Concepts Sapphire II. And for "air"—inflate it to 32 pounds, will you Kevin?—there's the rearward-firing, switchable Audax supertweeter that David doesn't mention.—Ed.]

It is unclear what caused my negative impression of the sound of the Snell Type Q. Perhaps the minimum-phase requirement is more important in speaker design than previously thought. *[Not bloody likely.—Ed.]* Another potential problem arising from the even-order Linkwitz-Riley filter is the nonconstant-power crossover [Vanderkooy 1986]. This results in a reverberant field which will not be flat in the crossover region. The effect should not be apparent in a home listening room, since the direct sound dominates what initially reaches the listener's ear [Dickason 1987]. In the final analysis it is probably just the speaker's spectral balance that caused me to react negatively to it. The fact that I found the balance of the Snell Q to be unsatisfactory should not deter you from considering the speaker. Of the speakers reviewed here, the Snell most closely follows the design approaches and design criteria outlined by the majority of published researchers in the field of speaker design. If you find the spectral balance of the speaker acceptable, then it can be recommended. *[I found the subjectively perceived spectral balance of the Type Q to be virtually identical to that of the Type C/IV, except for the bass, of course, which rolls off below 80 Hz at 12 dB per octave. See my C/IV review in this issue for specific comments.—Ed.]*

Infinity Modulus

Infinity Systems, Inc., a Harman International Company, 9409 Owensmouth Avenue, Chatsworth, CA 91311. Modulus compact 2-way loudspeaker system, $1000 the pair. Tested samples on loan from dealer.

This speaker was made available on a very brief dealer loan. It is much smaller than the speakers discussed above, with a woofer only 5 inches in diameter. That is the size of most midrange drivers. Given the small woofer diameter, the speaker must be crossed over to the subwoofer at a frequency of 150 Hz, or higher. A separate subwoofer for each channel would be required at that crossover frequency for optimum performance. Infinity claims that a single subwoofer can be used; they sell a powered subwoofer for use with the Modulus, at $2000.00. The cabinet of the Modulus is made of a nonresonant plastic compound. The tweeter is a small EMIT ribbon driver, custom designed by Infinity. A fourth-order Linkwitz-Riley crossover is used.

The speaker gives a very dynamic presentation, but it has a very unpleasant midrange coloration. At first exposure the speaker's dynamics are more apparent than the midrange coloration. Over a longer term, the midrange coloration becomes more bothersome. We observed that the speaker had lower harmonic distortion than the Spica and Audio Concepts. That might explain the increased sense of dynamics. But we also observed significant ringing in the tone burst and pulse tests. This correlates with the midrange coloration. We also found significant variations in the frequency response as the microphone was moved around the vertical axis of the speaker. A well-designed speaker that uses a Linkwitz-Riley crossover would not exhibit this problem.

At half the price, the Spica TC-50 provided a more satisfying listening experience. The Infinity Modulus is thus not recommended.

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Basic Issues of Equipment Reviewing and Critical Listening: Our Present Stance

By Peter Aczel
Editor and Publisher

For the benefit of new readers, as well as longtime readers who may need to be reminded, here are some of the fundamental viewpoints that divide responsible audio reviewers from the tweaks and cultists.

If you read a lot of audio publications and converse with a lot of audio people, as I do, you know that the line has been drawn between two opposing factions. The audio world is at loggerheads as never before. The so-called objectivists and subjectivists have evolved highly divergent belief systems; each side shows a basic lack of respect for the other; accusations of self-serving politics and defective hearing abound; the general tone is uncomfortably confrontational. In the heat of the arguments, science and logic are forgotten, methods and credentials are left unquestioned, obfuscation is rampant, and wimpy suggestions to the effect that the truth lies in between are slipped in sideways by the knee-jerk conciliators. This is a good time, indeed an obvious time, for ***The Audio Critic*** to restate its position on the issues that constitute the basis of the ongoing debate.

What sounds different?

To the dyed-in-the-wool subjectivists, everything sounds different. One of my favorite dirty tricks is to go through the motions of conducting a single-blind A/B amplifier or preamplifier comparison which is actually an A/A comparison because I only pretend to switch to B but never do. Lo and behold, some of the audiophiles in attendance claim to hear major differences in front-to-back depth, imaging, "air," etc., and are quite certain they can pick out A and B blind. A cruel experiment but educational. Thus I have no fear that such audiophiles will argue with me when I list the various elements in the audio chain that really *do* sound different. To wit:

Listening rooms—and how! Loudspeaker systems, even the relatively accurate ones. Surround-sound and other environment processors, obviously. Phono cartridges and tonearms, if you still care. Microphones—very important and all very different. Recording studios and concert halls, for the same reasons as listening rooms, only more so. And finally, the widely differing recording techniques of different record companies, producers, and engineers—even when they use the same microphones in the same hall. What else sounds different? That's just about all I can think of. (No, I'm not forgetting wires and cables. They constitute a very special case, subject to serious misrepresentations, and are treated separately in this issue.)

What sounds the same?

Here we come to highly divisive subject matter, the major source of hostilities and character assassinations in the high-end audio press, but there's no reason for rational audiophiles to doubt what has been demonstrated over and over again in properly conducted double-blind listening tests. Power amplifiers, preamplifiers, CD players, D/A processors, DAT recorders, FM tuners, and turntables sound the same—with certain very important qualifications.

What are those qualifications? Power amplifiers must have high input impedance, low output impedance, no frequency-response anomalies, and be at all times operated within their voltage and current capabilities in order to sound the same. Preamplifiers must likewise be without equalization errors, other frequency-response anomalies, and overload problems in order to sound the same. Digital audio equipment must be up to the present-day level of converter technology and, analogwise, meet the aforementioned preamplifier qualifications in order to sound the same. FM tuners will sound the same only when receiving a strong signal without multipath. Turntables will sound the same only if adequately isolated, damped, and free from drive irregularities. Without these qualifications all arguments on the subject are meaningless.

In general, any two components A and B that can be alternately switched into and out of an audio system in an A/B test will sound the same if (1) their linear characteristics are essentially identical and (2) their nonlinear characteristics are below the threshold of audibility. If you think about that statement for a minute, you begin to realize that it's a truism rather than a heresy; the trouble is that the tweaks and cultists often think for less than a minute.

Example: The biggest Krell power amplifier and the smallest Hafler are being A/B'd through a highly accurate but inefficient speaker system in a large room. The Krell is seven or eight times as powerful and about 37 times as costly as the Hafler—a totally ridiculous comparison, right? But if the program material is, say, a quiet, melancholy Spanish guitar solo, I guarantee that you'll hear no difference between the two amps if the test is correctly set up (see below) because at that level they're both perfectly flat, linear, and nondistorting. Now play the Saint-Saëns "Organ" symphony through the same lash-up and the little Hafler will begin to sound ugly (i.e., nonlinear) in the heavy passages, whereas the Krell will keep sounding gorgeous because of its almost unlimited voltage/current capability—and not because of some special high-end circuit features.

Same or different—how do you know for sure?

The only reliable way to determine whether two audio components sound the same or different is by means of a double-blind listening comparison at matched levels.

All other methods lack credibility (if indeed they can be called methods at all), but I happen to be extremely permissive when it comes to the specific rules of the double-blind test. I use the superbly convenient ABX Double Blind Comparator because it permits instant switching—not only between A and B but also between A and X or B and X (X being the randomized unknown, either A or B)—but if you're against rapid switching or believe that high-quality relays introduce colorations (they don't), go ahead and have someone you can't see or hear switch cables by hand, and take all the time in the world, say 16 weeks for 16 guesses, if you think that's better. There are only two unbreakable rules: A and B must be matched in level within 0.10 dB (you can get away with 0.15 dB but that's the limit), and there must be no clue other than the *sound* of A and B when you're ready to make a blind indentification. Everything else is negotiable.

I think I can explain why experienced and honest audiophiles insist that a certain power amplifier (or preamp or whatever) sounds better than another when I know from my own tests that the two are indistinguishable. Level is the key to that paradox. The human ear can detect level differences and changes as small as 0.2 dB across a wide range of frequencies. In fact, level totally dominates our perception of sound. The trouble is that when A and B differ in level by, say, 0.35 dB, we hear a difference all right, but we may not identify it as a difference in level per se—it may come off as a subtle difference in quality. A difference that everyone can unmistakably attribute to level is usually close to 1 dB. That's why level matching by ear in a listening comparison is a no-no; it just isn't accurate enough. You've got to use a meter—and most audiophiles don't have dB meters accurate to 0.1 dB. Now, as soon as there's any audible difference between A and B, even if due exclusively to level, audiophiles will tend to assert that one "blows away" the other. To the faithful, there exist no small differences.

I'll go further. Even without a side-by-side A/B comparison, level can play tricks on you. An audiophile brings home a new power amplifier with a gain of 27 dB, instead of 26 dB like his old one. (Never mind the difference, if any, in voltage/current capability.) He inserts his new acquisition into his system, leaves the volume control where he usually has it, and after two minutes starts raving about the night-and-day difference in imaging, depth, etc., etc. What he doesn't realize—and what I myself didn't realize for many years—is that listening level has a great deal to do with the subjectively perceived "personality" of the equipment and that truly accurate level matching, on the other hand, makes everything sound startlingly, *scarify* similar—even when the audibility of a small difference is validly established in the end (say, in the case of power amplifiers, because of a large difference in output impedance). I firmly believe that the conscientious level matchers will all end up on my side of the controversy on this subject, whereas the typical catch-as-catch-can listeners will continue to argue endlessly without resolution.

What claims can be taken seriously?

Today's audiophile is subjected to an unceasing stream of claims for new products promising to deliver dramatic improvements in sound quality. Some of these claims are accompanied by engineering rationales having varying degrees of credibility; others are exercises in mysticism, magic, greed, or just plain ignorance. As an audio journalist and publisher, I don't believe that anyone who raises his hand to announce that he has come up with a stupendous product is automatically entitled to a hearing (i.e., an eventual review). Life is too short to give "equal time" to highly qualified *and* totally unqualified practitioners—at least from the point of view of one graying editor. So, please, don't ask me how I know that this 100% pure platinum line cord doesn't make the amplifier sound better when I haven't even tested it. I'm busy testing other things.

What are my criteria for considering a new product worth the time and effort needed to test it? (1) It should come from a designer I respect, *or* (2) it should have an engineering rationale I can believe, even though I don't know the designer, *or* (3) it should claim some readily measurable superiority in performance, even if the engineering rationale isn't divulged, *or* (4) it should claim better sound, regardless of measurements, than is otherwise available, as verified in properly conducted double-blind listening tests. The mere assertion that something "sounds better" to your golden ear than anything else doesn't cut it with me, or my associates, anymore. Don't assert it; prove it. Pick it out 12 or 13 times out of 16 in a double-blind test. Or show me something measurable that *might* make it sound better to an exceptional ear, not necessarily mine. Show me *something* —anything—but don't just give me that golden-ear routine.

Speaking of exceptional ears, I want to reiterate my long-standing belief that if one listener out of a hundred—or even one out of a thousand—can actually hear a difference between A and B under the controlled conditions discussed above, then it's a real difference and worth every consideration by audio professionals. Saying you can hear it isn't the same as actually hearing it, however. Reviewers like Anthony Cordesman, Martin Colloms, Harry Pearson, Dick Olsher, etc., may say that they hear a difference in forwardness, or recessiveness, or warmth, or speed, or whatever, between solid-state amplifiers A and B, but I'm willing to bet the ranch that they can't hear it in a double-blind test at matched levels. Of course, they probably wouldn't submit to the test. Self-indulgent, nonaccountable, solipsistic expertizing is more appealing to certain minds than mundane reality. As I've pointed out before, the more expertly detailed and quasi-pornographically explicit the description of a *linear* electronic signal path's sound is, the more I suspect the writer of being a.. .er.. .sphincterated caudal orifice.

Where does all that leave us?

I realize that some simple souls will now say that ***The Audio Critic*** has become one of those every thing-sounds - the-same hi-fi magazines. That's nonsense, of course (and, in view of what's actually written above, indicative of a reading problem), but let me remind all audiophiles that equipment reviewing isn't—or at least shouldn't be—just a matter of A-sounds-better-than-B. Suppose A is found to sound indistinguishable from B in a careful ABX comparison. I might still recommend A over B because it measures better—and, who knows, with future software, or future ancillary hardware, or to a one-in-a-hundred golden ear, it could end up sounding better. Also, if A is better built than B and promises to last longer, it should definitely get the nod. Of course, if A is priced at $6000 and B at $1200, *and* they sound the same, the better measurements and better construction begin to look a lot less attractive, so I would in all likelihood advise against A. If, however, the same A is only $200 or $300 costlier than B, I would almost surely say that the better measurements and construction are worth the difference.

Far more important than any such nit-picky matters are the true first-order differences in audio equipment, such as for example the wave-launch characteristics of different loudspeaker designs or the bass capabilities of different amplifier/woofer/room combinations. First things first—the first-order effects before the second-order nits—should be the guideline of intelligent audio journalism, but that's not what I see out there these days. I find it very sad, and unendingly frustrating, that a whole publishing industry and an entrenched consumer cult have grown out of the obsession with second-order effects, and that only a few academics and professionals insist on the logical priorities. Can you imagine how much better the typical audiophile's stereo system would sound if all the garbage about silver cables, tiptoes, CD rings, custom line cords, "tube sound," etc., had never been written and if, instead, the touted cult items were inexpensive high-powered amplifiers (such as Adcom or B&K or Carver) and big woofers? The mind boggles.

Look to ***The Audio Critic*** for a corrective approach to this depressing situation. Maybe, in the end, reality will prove to be as appealing as fantasy. 0

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The Electronic Browsing Section: A Collection of Totally Unrelated Pieces of Audio and Video Equipment

By Peter Aczel
Editor and Publisher

Sorry, there are too many different categories here and only one or two reviews in each, so our usual survey type of article is out of the question. Still, each separate item is definitely worth your attention.

If you've read the preceding article, you know why you won't find voluptuous descriptions of the midrange of an amplifier, the imaging of a CD player, etc., in the reviews that follow. A normally operating amplifier has no characteristic midrange distinguishing it from others, and a normally operating CD player images according to the dictates of the recording, not the circuit board. As long as such matters are understood, we can proceed.

Metal Stand for the Quad ESL-63
Arcici Q-l

Arcici, Inc., P.O. Box 1502, Ansonia Station, New York, NY 10023. Quad Stand Q-l with "Super Spikes," $250.00 the pair. Tested samples on loan from manufacturer.

In my Quad ESL-63 review in Issue No. 14, I stated that I'm still looking for the ideal stand for this unique electrostatic speaker. Well, the Arcici may or may not be ideal, but it works pretty much as it's supposed to and, besides, it's the only game in town. It's a good, solid stand that makes the speaker almost 50 inches tall and raises the virtual point source of the wave launch to somewhere near ear level. That's the way the ESL-63 sounds best, and I don't know of another stand that does the same, so the bottom line is that you've got to have it if you're a Quad user.

The design is very simple, maybe too simple, but it does the job. To illustrate it typographically, it's basically nothing more than this: . Imagine four such dinguses made of flat rectangular steel pipe, two for each speaker unit. That's the kit you get, along with spikes, hardware, etc. One side of the speaker fits between the two upright pieces, held in place with sharp set screws; the end of the bottom panel is held by the horizontal angle iron; and the same thing happens on the other side. The support on each side is independent; there's no connecting piece. The arrangement is rock solid, as long as you don't have to move the speakers—but that's exactly what a reviewer has to do all the time. The trick then is to grab the Quad by the top wood panel and the bottom assembly, and not touch the Arcici stand at all, otherwise you might shear the set-screw-held uprights away from the speaker. Ugh. A crosspiece connecting the two sides would have solved this problem. I must say, however, that if you don't use the stand as a handle, the set screws will never loosen. I try to tighten them from time to time because I don't trust them, but they never need tightening. Son of a gun.

The hollow uprights can be filled with sand or lead shot to make the stands heavier and deader. I have nothing against stands that are nice and heavy and dead, except that I don't like to move them—especially with that awkward handhold—so I left the uprights hollow. As for the spikes that come with the kit, use them if the footing is unstable; in my listening room it happens to be very stable. Tweaks and cultists who use spikes under any and all conditions because they believe the speakers will "sound better" deserve all the extra inconvenience that ensues. Overall, I think the Arcici stand will make most Quad people quite happy. Pricewise it's no bargain, but neither is the ESL-63.

Electronic Crossover
Bryston 10B

Bryston Ltd., 57 Westmore Drive, Rexdale, Ont., Canada M9V 3Y6. Model 10B Active Crossover, $1095.00. Tested sample on loan from manufacturer.

This is a beautifully engineered, electronically flawless piece of equipment of limited usefulness. Crossing a separately amplified subwoofer over to the main speaker would be one of its more obvious applications; more about that in a moment. Here's what the 10B can do—and anything it can do, it really does perfectly.

In each channel, it can select 12 crossover frequencies, more or less evenly spaced between 70 Hz and 4.5 kHz, and activate Butterworth lowpass and highpass filters that have the selected frequency as their passband edge. The attenuation slopes of the lowpass and highpass filters are separately adjustable to 6, 12, or 18 dB per octave (1st, 2nd, or 3rd order), and the highpass filter level as referenced to the fixed lowpass filter level can be set in 1 dB steps from -5 dB to +5 dB. And that's not all, as they say in those special offers on TV. By manipulating connections on the back panel, you can turn the 10B into a *mono* crossover of even greater versatility—would you believe a variable-slope three-way or a 4th-order Linkwitz-Riley two-way?—but then of course you'll need two units for a stereo system. There are also professional versions with balanced inputs and outputs, special Linkwitz-Riley modules, you name it—hog heaven for the biamp/triamp crowd.

My measurements revealed absolutely no flaws, errors, or glitches in this complex system; the filter contours that 1 checked at random among the available permutations and combinations were all dead-on; distortion and noise were pretty nearly unmeasurable on my test bench at all audio frequencies regardless of the filter settings; in other words, the signal paths of the device appear to be perfect. (All right, there *is* one potential—but easily remediable—problem. Inside the unit, a 10-ohm resistor between chassis ground and signal ground appeared to be the cause of a slight but audible hum in the biamped system of one of my associates. Shorting the ground side of any one of the output jacks to the chassis killed the hum.)

David Rich, whose various EE degrees also stand for El Exigente, had only good things to say about the circuit design, which is implemented with discrete op amps. He praised the elegant simplicity of various engineering solutions in the 10B and called designer Chris Russell "a ridiculously good engineer," by which I think he meant that Chris goes to almost ridiculous lengths to refine his circuits and minimize distortion, without allowing the cost-effectiveness of his designs to go down the drain. That's what good engineering is all about.

As for the limitations of the 10B, they have nothing to do with engineering but stem from the basic problems of crossing over real-world drivers, which are very different from the idealized amplifier loads assumed by a "perfect" electronic crossover. Real-world drivers are, in effect, lowpass and highpass filters; only a dedicated crossover, whether passive or active, can process those filter characteristics in such a way that the interacting electrical/acoustical poles and zeros will yield the combined, measurable lowpass and highpass responses required in a particular design. In other words, a truly good crossover for a specific speaker system can't be separately bought off the shelf. The exception to that rule would be a subwoofer crossed over well below its upper roll-off frequency to a more or less full-range main speaker system. That way there are no preexistent poles imposed on the electronic crossover in the vicinity of the crossover frequency. Bryston has also come to the realization that this is the best possible use of the 1 OB and has recently added a new model, the "l0B-sub," to the line, with all 12 crossover points at lower frequencies (40, 50, 60, 70, 80, 90, 100, 200, 250, 300, 400, and 500 Hz). I think that makes a lot of sense.

As a subwoofer crossover, the 10B is unquestionably state-of-the-art and very reasonably priced for such a complex piece of equipment. I see no point in evaluating it subjectively, since the perceived sound quality will depend entirely on the speakers used and on the specific settings of the controls; the electronic signal path as such is obviously transparent. If your biamped subwoofer setup requires, let us say, 18 dB per octave Butterworth filters crossed over at 100 Hz for best results, you can be certain that no better solution exists than the Bryston 10B. And if you then decide that 70 Hz would be a better choice, the changeover will be totally painless. But don't imagine that you're a crossover designer for 2-way and 3-way speaker systems just because you own a 10B. There's a little more to it than that.

Professional Power Amplifier

Carver Model PT-1250

Carver Corporation, P.O. Box 1237, Lynnwood, WA 98046. Professional Model PT-1250 "Pro Touring Magnetic Field Power Amplifier," $1500.00. Tested sample on loan from manufacturer.

This is not part of the Carver consumer amplifier line familiar to audiophiles; it's sold mainly to rock groups through professional distributors. I find it more intriguing however, than the Silver Seven-t, TFM-45, TFM-42, etc., for a number of reasons.

For one thing, this incredibly powerful stereo amplifier, conservatively rated at 465/465 watts into 8 ohms and 625/625 watts into 4 ohms, weighs 10 pounds. That's not a typo. Ten pounds—one oh. That's why rock groups like it; they can take a whole stack of them on the road without any weight penalty and have all the wattage they'll ever need. The Carver magnetic-field power supply reaches its ultimate state of refinement here in terms of watts per pound; it must be some kind of world record. Part of the secret, however, lies in the monocoque construction of the chassis, borrowed from aerospace technology: the outer "skin" carries all of the stresses, and when the chassis is opened up, the thin metal actually goes limp, although with the screws in place the whole assembly is perfectly rigid. It's a fascinating tour de force and of course a welcome relief from the brink-of-hernia syndrome incurred when installing or moving other superpowered amplifiers.

All that would be no more than an industrial curiosity if the PT-1250 weren't a perfectly clean, audiophile-quality amplifier. In fact, I prefer its classic solid-state transfer function to what Bob Carver does in his razzle-dazzle tube clones, nice as they are. The damping factor of the PT-1250 at 1 kHz is rated at 200, as against 7 or 8 in the case of the Silver Seven and its clones, with their deliberately high output impedance. The professionals obviously want the output into a real-world load to be a replica of the input, without any tubelike processing. (This is an ongoing and unresolved debate between me and Bob.) I also appreciate the balanced inputs of the PT-1250, manadatory in a professional amplifier but advantageous in any installation. The one thing that might keep the PT-1250 from acquiring an audiophile following is a somewhat noisy fan, needed in the absence of sufficient metal to act as a heat sink. The fan has two speeds, but even at the lower speed it's distinctly audible in a quiet room. At a rock concert that's not an issue; in a home installation it could be a marginal annoyance.

The fan noise was the main reason why I didn't do a full-fledged ABX comparison between the PT-1250 and my laboratory pair of Boulder 500AE's in their mono-bridged mode, rated at 500 watts into 8 ohms. Despite the significant discrepancies in measurable distortion—the Boulder has an extra zero after the decimal point and then some—the two sounded so much alike at matched levels that the fan noise would certainly have masked whatever minuscule differences one might (or might not) have detected double-blind against a completely silent background. I lived with the PT-1250 in place of the Boulders for over a week (if you'll pardon the anecdotal evidence) and was basically just as happy, except for occasional Hungarian maledictions when the fan covered up a *ppp* passage. Don't misunderstand me; I'm used to a very quiet room, but a lot of people have a higher ambient noise level in their listening room than the low-speed noise of the fan (not to mention their noisy LP surfaces). A far more important point is this: given a basically intelligent circuit design, the more powerful amplifier is the better amplifier for music, and the Carver PT-1250 gives you all the power in the world without the usual penalties in size, weight, and price. (I'd much rather own the PT-1250 than, say, the 100/100-watt Mark Levinson at two and a half times the price—even if I didn't have to pay for either—because orchestral climaxes, fortissimo high C's, drum solos, etc., will be more natural-sounding through the more powerful amplifier, whereas at lower levels there's every reason to expect them to sound the same. "If this be treason, make the most of it.")

The Carver magnetic-field power-amplifier topology makes a minor sacrifice in ultimate distortion figures (single-oh instead of double-oh percentages) in order to achieve the lowest possible size/weight-per-watt figures, although even that trade-off could be eliminated with extraordinary measures that Bob Carver considers overkill. As it is, the PT-1250 acquitted itself very satisfactorily on the lab bench THD-wise, but its long suit is of course power. Maximum single-channel continuous power output at 1 kHz is approximately 670 watts into 8 ohms, 990 watts into 4 ohms, and 620 watts into 2 ohms. The dynamic (old IHF-type) power readings are only a few watts higher than those numbers, indicating almost perfect power supply regulation. With both channels driven simultaneously, maximum continuous power output at 1 kHz is approximately 550/550 watts into 8 ohms, 805/805 watts into 4 ohms, and 578/578 watts into 2 ohms. Furthermore, one channel can "borrow" the unused power capability of the other, up to the single-channel capability; thus, as an extreme example, 670/430 watts into 8 ohms instead of 550/550 watts is a possibility. As I said, the official ratings are very conservative. THD at the rated 465 watts into 8 ohms ranges from 0.0085% at 20 Hz, through 0.045% at 1 kHz, to 0.2% at 10 kHz, then drops back to 0.07% at 20 kHz. At the rated 625 watts into 4 ohms, the corresponding distortion figures are approximately twice as high. The measured damping factor, as referred to 8 ohms (i.e., 8 ohms divided by the output impedance), is approximately 280 at the lowest frequencies, 180 at 1 kHz, 100 at 2 kHz, then declines linearly to 14 at 20 kHz. That's still tantamount to a voltage source. As for bandwidth, the small-signal (1-watt) response is down 1 dB at 10 Hz and 0.8 dB at 50 kHz.

Overall, the Carver PT-1250 is a reference-quality power amplifier even if you disregard its size, weight, and price—except for two things. One is that fan, the annoyance factor of which depends on placement and surroundings. The other is the absence of super-duper specs at the highest frequencies, which is of importance only in laboratory work, not in a music system. Besides, you just can't disregard this particular amplifier's size, weight, and price.

Preamplifier
Coda 01

Coda Technologies, Inc., 9233 Wausau Way, Sacramento, CA 95826. FET Preamplifier 01, $2500.00. Tested sample on loan from manufacturer.

I can't get terribly excited over a new preamp these days, even when it's a really gorgeous piece of hardware like this one. A modern preamp is about as predictable as vodka (whereas a modern loudspeaker is only as predictable as scotch); voltage gain and equalization are a pretty exact science these days. I therefore passed the buck to David Rich—he is younger than I and not as jaded. He also knows a hell of a lot more about electronic circuitry, and he studied the circuit diagram of the Coda 01 as if it had been handed in as an assignment by one of his EE students; then he called the designers and asked a lot of questions. His comments follow below, after the three asterisks. All I did was this:

I examined the low-silhouette, machined-metal, all-anodized chassis, took off the cover, and marveled at the construction and parts quality. The cliche phrase would be "a work of art," but in this case it's no cliché. No integrated circuits, discrete semiconductors only (mostly FETs), no electrolytics except in the power supply, gold-plated circuit boards, superb parts layout, nothing but the best—and the best-looking. My only criticism has to do with those long cylindrical control knobs; they're nice to look at but hard to read for index position. Not practical.

Then I took some measurements and verified that the Coda 01 meets its highly respectable distortion, noise, frequency response, and RIAA equalization specs with margin to spare. Finally, I inserted the unit into my own system—

balanced outputs are available, just as on my Boulder MS—and listened. Everything sounded perfect, just as I expected. I also did a little ABX-ing against the Boulder, or rather just A/B-ing, as I could hear no difference between fully identified A and fully indentified B—so why try X? Again, just as I expected. I'll let David Rich continue. *[Ed. stops here.]*

* * *

The Coda 01 belongs to a class of modern preamps characterized by a number of common design approaches. Other members of the class include preamps from Aragon, Bryston, Krell, PS Audio (5.5), and Boulder. The lowest-priced unit in the group is the Bryston .5B at $750.00. These preamps are characterized by the use of discrete active stages and a two-stage phono equalizer. The discrete topology allows the design of a phono stage with lower noise than could be achieved with a single IC. The two-stage equalizer has several advantages: (a) the open-loop gain requirement for each operational amplifier is reduced, (b) the overload characteristics of the phono stage can be improved, (c) loading effects of the RIAA equalization network on the operational amplifier can be reduced, and (d) the 6-dB-per-octave rolloff of the RIAA equalization curve can continue above 40 kHz.

The output sections of all these preamps are class A with a quiescent current on the order of 20 mA. The worst-case current that these preamps will source and sink into a single-ended load is an order of magnitude below this quiescent current. The open-loop distortion from the output stage of these preamps is thus very low. This is in contrast to integrated circuits, which operate in the class AB mode with low quiescent currents. I/*V* current limiting, which is incorporated in integrated circuits, is not present in the discrete designs.

In discrete design, compensation techniques to prevent oscillation when the feedback loop is closed can differ significantly from those used in integrated circuit designs. The compensation methods differ because the discrete designs are application-specific, in contrast to an IC which must work in hundreds of different applications. The open-loop transfer function of the gain stages in these preamps is kept relatively constant in the audio band through the use of these compensation techniques. As a result, distortion is kept at vanishingly small levels (below -86 dB on 6 V peak-to-peak signals) throughout the audio band.

There are significant differences in discrete op amp circuit topologies used in these preamps. In the case of the Coda 01, a differential pair is used in the second gain stage as well as the first gain stage of the discrete operational amplifier. N-channel JFETs are used in the differential pair of the first stage; p-channel MOSFETs are used in the second-stage differential pair. Bipolar devices are used in all other functions, including the output stage. The differential-to-single-ended conversion is performed by an active current mirror, which is used as an active load on the second differential stage. Very high common-mode rejection ratios are achieved using this topology. Each MOSFET in the second gain stage is connected to a common-base bipolar (cascode) stage to improve open-loop linearity.

The phono stage is capacitively coupled. The coupling capacitors are part of a second-order subsonic filter (14 Hz) which cannot be bypassed. The RIAA equalization is performed actively. The 2122 Hz pole is synthesized in the first stage. The 50 Hz pole and 500 Hz zero are synthesized in the second stage. The line stage is direct-coupled. DC is nulled in the line stage by the use of a trim pot. The output followers of the line stage are outside the feedback loop. This improves stability of the line stage when driving capacitive loads, but at the cost of increased output impedance and distortion. The tape monitor outputs are buffered. This prevents a powered-down tape recorder from loading the signal source.

Open-loop bipolar pass transistors are used as voltage regulators. The base of each of the pass transistors is connected to a filtered zener diode reference. Coda chose to use an open-loop regulator because it is unconditionally stable. The power-supply rejection ratio of an open-loop regulator is significantly poorer than that of a regulator which uses feedback. Separate regulators are used for the left and right channels. A total of 10 regulators is used.

Many high-end preamps omit the output muting relay on the grounds that it colors the sound. *[There we go again.—Ed.]* With such preamps a power supply interruption will result in a large turn-on transient that can destroy your power amplifier and speakers. The Coda 01 also does not use a muting relay, but it is eliminated to improve reliability. According to Coda, the muting circuit is one of the circuits most likely to fail in a preamp. The active circuitry of the Coda 01 is stable on power up. Accordingly, it will not emit pulses on power up.

The Coda 01 has a separate Record Selector switch. This allows one source to be monitored while another is recorded. It also allows tape-to-tape copying. The design is flawed because the two tape monitor outputs are connected together. If the Input Selector and Record Selector are both set to Record One (or Record Two), a potential destructive oscillation can occur. The problem will occur if any tape recorder is set to monitor the source signal. The oscillation results because the input and output of the tape recorder become shorted together. Competitive products, such as the Bryston 11B, do not suffer from this design flaw. In the Bryston, separate tape outputs are provided for each tape monitor loop. Separate selector switches are included for each tape recorder on the record selector assembly. The tape monitor 1 output cannot be connected to the tape monitor 1 input signal.

If your taste runs to Rolex watches and Leica cameras, then the Coda 01 preamp is for you. It offers a superior look and feel in comparison with less expensive products. But equivalent performance and reliability can be had for one third the price of this preamp.

—David Rich

[Ed. resumes here.]

CD Player Mod
EAD "AccuLinear"

Enlightened Audio Designs Corp., 508 North 2nd Street, Fairfield, IA 52556. EAD Ultra CD Player Mod, $599.00; EAD Premiere CD Player Mod, $399.00; new Rotel RCD-855 with Ultra Mod, $899.00; new Rotel RCD-855 with Premiere Mod, $699.00. All mods direct from EAD. Tested samples on loan from manufacturer.

I think this small but obviously smart young company is on to something, maybe not something very big but something indisputably valid, unlike so many voodoo-based little audiophile operations. The basic insight that informs the company's efforts is that the Achilles' heel of present-day digital playback is not in the digital domain but in the current-to-voltage (I-to-V) conversion stage after the DAC. (Some of the pitfalls of this stage and the applicable solutions were discussed by David Rich in the sidebar on pp. 24-25 of Issue No. 15.) Enlightened Audio Designs has a proprietary circuit called AccuLinear, which is claimed to perform the I-*to-V* conversion at the output of a multibit DAC with lower transient distortion and noise—including ultrasonic and RF noise—than anything else on the market, thanks to (as I understand it) less slewing and faster settling.

Unfortunately, EAD offers no test at the audio ouput of the CD player to prove these claims; the measurement they claim to have improved is at a pinout of the DAC chip (the summing junction), which in the particular implementation they had sent me was not accessible without highly invasive maneuvers. No matter; their point is well-taken, and I believe their measurements. Others have made similar claims—for example, MSB Technology—and it's possible that the whole big "proprietary" deal amounts to nothing more than a high-speed op amp, but that's all right with me, too. It's the correct approach to an important detail, and very few are doing it that way.

The unit sent to me for testing was the Philips-based Rotel RCD-855 with the top-of-the-line AccuLinear mod. David Rich recommended the 855 in his article in Issue No. 15, and at $349.00 (list price) the unmodified player is indeed an excellent buy. At the direct-from-EAD prices, the Rotel mods are no longer good buys because, regardless of the more sophisticated circuitry, you're still getting a rather austere, entry-level CD player in terms of construction and features. EAD considers the 855 to be a nice, solid, trouble-free platform for their mods, but in my opinion the package is unrealistically priced. To name just one shortcoming, there's no index search facility. To a classical CD collector, that's the kiss of death. Don't give up on EAD, however; I'll give you some good news in a moment.

I put the player through my usual lab-bench measurements and determined that it was a typical, good Philips 16-bit machine. That means low-level gain-linearity errors of the order of ½ LSB and just a smidgen of harmonic distortion visible above the noise floor at those levels—better than what I found in the high-end Philips LHH500 but nothing exceptional. I did see a little bit of RF at the output—up to 100 MHz, modulated in envelopes of 25 to 30 mV peak-to-peak amplitude—despite the EAD claims. Of course, RF has a way of sneaking in sideways; the EAD circuitry wasn't necessarily at fault.

As for the sound, I can't confirm EAD's rhapsodies of "much cleaner, more dynamic sound...with better sound-staging, musical clarity, and inner detail," nor can I take seriously the subjectivistic raves without proof that they quote from one of the undisciplined, self-indulgent underground journals. No, sir, but I can report something that should be more impressive than all that silliness even to the EAD people themselves: I set up a very careful ABX listening comparison between their Rotel mod and the Theta DS Pre Basic—the most sophisticated piece of D/A converter equipment I had on the premises, fed from the coax digital output of the Rotel—and I could hear no difference between the two on a variety of program material. That's an objectively structured and controlled subjective experiment, not a restaurant-review type of exercise in exquisite personal taste.

Now for the good news I promised. EAD is about to come out with a really handsome outboard D/A converter box that combines their AccuLinear circuitry with one of the new state-of-the-art 20-bit DACs (Analog Devices AD1862N- J) and a very respectable digital filter (NPC SM5813ATT), plus a first-rate power supply and neat little features like a phase inversion switch, de-emphasis indicator, decoding error indicator, etc. The projected price is $1299.00 (retail, not direct). When I was told about this product, my reaction was—now you're talking! It's the right package for the kind of technology a small company like EAD has to sell, and the price seems to be reasonably competitive. Don't forget, almost any disc transport will do if the EAD box is correctly designed. I may even want one for myself. We shall see.

CD Drive Unit and Multi D/A Converter
Esoteric P-2 and D-2

TEAC America, Inc., 7733 Telegraph Road, Montebello, CA 90640. Esoteric P-2 Drive Unit, $4000.00; Esoteric D-2 Multi D/A Converter, $4000.00; RC-356 Remote Control Unit included. Tested samples on loan from manufacturer.

Here it is: the ultimate Japanese statement on digital playback in general and CD reproduction in particular—at least in intent. As such, it competes with the top-of-the-line Wadias and Krells, the now retired Sony CDP-Rl-cum-DAS-R1, and others in that exalted category. Let me state right up front that the Esoteric P-2/D-2 isn't my cup of sake, but that doesn't necessarily mean it's not as good as claimed.

Cosmetically, the matching two-chassis set is quite striking with its massively sculptured look and highly textured, pink-gold metal and Nextel surfaces, although I think the appeal is more to the big-cuff-links-and-pinky-ring taste than to the more austere high-tech sensibility. The construction appears to be of the highest quality, although I stopped trying to open the Chinese-puzzle-like, deeper-than-wide boxes when forcing would have been the next obvious step. (The promised service manuals never arrived.) All I know—just on the basis of readily available information—is that the disc transport in the P-2 is of an extremely elaborate design, probably the most ambitious mechanical engineering project of its kind seen so far, and that the D-2 uses the Sony CXD1244 digital filter chip (good), two Burr-Brown 18-bit PCM 170IP DAC chips per channel (good), and a number of NJM5532 op amps for audio (not so good). That's basically just standard Japanese decoder circuitry with a few extras thrown in (optional balanced output, digitally controlled output level, polarity inversion, etc.); the Theta DS Pro Basic, for example, is a conceptually more sophisticated decoder at half the price. The basic tests I ran were approached, in any event, strictly on a black-box basis.

A word about CD transports, before the test results. Many audio journalists and equipment reviewers fall into the trap of seeking analog virtues (precision, mass, etc.) in a CD transport, as if it were a turntable. The digital facts of life are that, as long as the laser can read a zero as a zero and a one as a one, the accuracy of the decoded/reconstructed music signal, in both the frequency and the time domain, will depend entirely on the electronic circuitry used—and that's where the money should be spent. Thus the Esoteric P-2 drive unit, impressive as it is as a piece of audiophile jewelry and as an exercise in precision mechanics, doesn't get you one iota closer to audio perfection than a nice standard transport assembly like the Sony G chassis. Those who can "hear" the difference between the P-2 and others will have to report authenticated double-blind comparison tests before I'll take them seriously.

On the lab bench, the D-2 (fed by the P-2, of course) exhibited perfect gain linearity at all levels, including the lowest, and no harmonic distortion blips whatsoever sticking out of the noise floor. That's good enough for me. On the other hand, I measured de-emphasis errors of the order of +0.2 dB (just a tad on the bright side) in both channels. I also found a major screwup, which may or may not be common to all samples of the D-2. When you press the Phase button and the LED indicator in the button lights up, the phase is *not* inverted as the instruction manual claims. The phase is inverted when the button is *not* pressed and the light is out. Pressing the button will *undo* the inversion—just the opposite of what's supposed to happen. This is unconscionable in a $4000 piece of equipment, but once you know about it, you can deal with it and use it correctly.

Ergonomically I found the P-2/D-2 combination to be OK but not great. The tray action, buttons and switches, display, etc., are more to my liking on various Onkyos, Pioneers, and Sonys, although there's nothing directly wrong with the Esoteric units. I wish there were a Stop button, not just Pause, on the front panel of the P-2. Stopping the transport without opening the tray is possible only by remote control, and that isn't always convenient. Personally, I'm much happier with CD players that have a more or less full set of controls on the front panel, so that you can misplace the remote control and still have a functioning player. That's not the way of The High End, however.

Since the appeal of the P-2/D-2 combination is basically visual and/or mystical and/or socioeconomic, and since I could discern no specific emphasis on *audio* perfectionism in the design concept, I decided to dispense with the usual ABX listening comparisons. I've had enough experience with CD players by now to know that there exists no scientific mechanism whereby the Esoteric equipment could have sounded better or worse than other good units (except perhaps when playing a very few pre-emphasized CDs). I was satisfied in the course of many weeks of ordinary listening that the sound was flawless in every respect.

Overall, I see no reason why anyone who is interested primarily in *results* should spend $8000 to own the Esoteric P-2 and D-2. On the other hand, if that kind of money is small change to you, you may very possibly enjoy the experience. Why not? What Dudley Moore said about life on a yacht in the movie *Arthur* certainly applies to the P-2/D-2 combination: "Well, it doesn't suck."

Compact Disc Player
Philips LHH500

Philips Consumer Electronics Company, One Philips Drive, P.O. Box 14810, Knoxville, TN 37914-1810. LHH500 Reference Series compact disc player with remote control, $2000.00. Tested sample on loan from manufacturer.

Once again, I'm reviewing a CD player at the tail end of its retail life. This one is still listed as current; in fact, it's more or less the flagship of the Philips line, the LHH1000 being no longer available; however, all Philips-made high-end equipment will from now on be sold under the Marantz label (as it has been in Europe), and the LHH500 will be quickly phased out.

All that would be more of a reviewer's dilemma if I could muster any enthusiasm for this high-priced product, but I just can't. I have considerable respect for Philips technology—only a fool wouldn't—but the Bitstream LHH500 isn't one of their best efforts.

Structurally and cosmetically, the unit is a one-piece version of the two-piece LHH1000, very solidly built on a die-cast aluminum alloy chassis, with a gold finish that intends to look expensive. Control functions and ergonomics are first-class, as long as you're willing to use the remote control even when the main unit is at your elbow, since only the most basic buttons are duplicated on the front panel. What's disappointing about the LHH500 is the circuit design and the Bitstream DAC's measurable performance. The discontinued Philips CD-80, costing 60% less, is actually a more desirable piece of equipment.

The master-slave power supply of the CD-80 is missing from the LHH500, and in place of the DC servo of the CD-80 we find an electrolytic capacitor, not even film-bypassed. (At $2000!) The differential to single-ended converter is similar to the circuit used in Sony 1-bit CD players. This circuit requires the op amp in it to reject significant amounts of high-frequency common-mode signals. The NJM5534 op amp used in the Philips LHH500's differential to single-ended converter doesn't have good common-mode rejection at high frequencies.

The Bitstream DAC system in the LHH500 doesn't even come close to state-of-the-art performance. I measured -1 dB gain linearity error at the -80.77 dB level in the right channel and -4.7 dB error at the -90.31 dB level. The left channel was only a tiny fraction better. That means we're just about down to 15-bit resolution, which is obtainable with almost any cheap player. My THD readings confirmed the low-level nonlinearities.

The DAC chip used in the LHH500 is the SAA7321, which was not designed for high-end applications and creates the need for an expensive kluge to implement the complete DAC circuit. The SAA7220 digital filter (also used in the 16-bit CD-80) replaces the internal digital filter of the SAA7321 to reduce the passband frequency ripple at the output of the latter. To achieve fully differential operation, fourteen SSI chips are used in conjunction with two SAA7321 chips. This is an expensive method to obtain a fully differential DAC output and contributes to the high cost of the LHH500, which doesn't seem to be justified by the measurable results. To say something positive, on the other hand, this is one of the best units I've tested as far as RF is concerned: there was hardly any coming out of the left channel and none out of the right channel.

The relatively poor value and impending obsolescence of the LHH500 gave me very little motivation to agonize over it in elaborate ABX listening tests. Let me just say that, on music, I noticed no difference between it and other expensive players. It appears that 15-bit resolution doesn't sound obviously faulty.

Now for the good news. The LHH500 will be superseded by a Marantz player in which the Bitstream DAC will be the SAA7350. This chip, which represents a significant advancement over the SAA7321, was discussed in the article by David Rich in Issue No. 15. Since then, Philips has introduced an important new chip to be used in conjunction with SAA7350. The new chip is the TDA1547; its function is to replace the analog section of the SAA7350. As explained in Issue No. 15, the analog section of the SAA7350 was compromised by the CMOS processing technology used to manufacture the chip. The TDA1547 performs all of the analog signal processing formerly performed within the SAA7350, but it uses a ±5 V power supply and an advanced analog process that combines *npn* bipolar transistors with complementary MOS devices. True 18-bit performance is claimed when the TDA1547 is used in conjunction with the SAA7350, and the combination—which will be used in the new Marantz player—promises to be competitive (and then some) with the top-of-the-line offerings from Burr-Brown, Analog Devices, and Sony. David Rich expressed to me the hope that the analog section of the new CD player will be designed by the same team that did the CD-80 and not the group responsible for the LHH500. If you can't wait to buy a Philips-designed CD player, the CD-80 at a closeout price is a possibility worth looking into.

Digital Audio Tape Deck
Sony DTC-87ES

Sony Corporation of America, Sony Drive, Park Ridge, NJ 07656. DTC-87ES Digital Audio Tape Deck, $1800.00. Tested sample on loan from manufacturer.

For a change, I'm not the last to review an important piece of Sony digital equipment. I may even be the first in this country, having received a very early sample, fresh from the introductory photo session.

This is Sony's top-of-the-line DAT deck for the consumer. That may be a borderline oxymoron, since the industry as a whole has just about given up on the future of the DAT format in the consumer market, as distinct from the professional market. Sony seems to be the principal exception, and the DTC-87ES is definitely a consumer-oriented product, bristling with "bells and whistles." Its main claim to fame is its four-head configuration, allowing the user to monitor the recorded tape (not just the source) during recording, just as if a three-head analog tape deck were being used. The four heads are the standard two for recording and playback, plus two additional ones for aftermonitoring. There are also four direct-drive motors—for the drum, the capstan, and the two reels in the DAT cassette—to ensure silent and stable tape transport. Quite a high-tech machine, although perhaps not as beautifully constructed as the less sophisticated Onkyo Integra DT-7700 I reviewed in Issue No. 12. That one was untouched by the RIAA-DAT wars; the Sony incorporates the compromise Serial Copy Management System (SCMS), which is quite unlikely to cramp the style of the audiophile type of consumer.

The circuit design of the DTC-87ES is typically Sony in that the analog circuitry is routine while the digital circuitry is pretty much state-of-the-art. The active gain elements in the analog record/playback stages are NE5532 and LF412 op amps, and the DC blocking capacitors in the signal path are—you guessed it—electrolytic. Not very impressive, but probably of no consequence as far as the audible results are concerned. At least the inscrutable decision makers at Sony opted for full ±15 V power-supply rails to put into their $1800 machine, so let's be thankful for that. The A/D converter is the Crystal CS5326-KP, a somewhat aging 1-bit design that was probably the best available when the DTC-87ES was designed. (Converters with better linearity are about to become available, including a very promising one designed by Bob Adams for Analog Devices. We'll keep you posted.) The 1-bit DAC is the excellent Sony

CXD2552; the 8 times interpolating digital filter that works in conjunction with the latter is the NPC SM5813APT. The inputs and outputs are Line In (L/R), Line Out (L/R), Digital In (Coax/Optical), Digital Out (Coax/Optical). Thus the deck is ready for any kind of outboard encoder/decoder box of the future, some of which will surely be even more high-end and high-performance oriented than the Sony circuitry, so that the deck will be usable as a tape transport/control center capable of receiving and outputting S/PDIF signals. Meanwhile the DTC-87ES is probably the best game in town for the audiophile who must have DAT now. At least I don't know of anything better.

I haven't used this machine extensively enough as a music-recording and playback device to have much to say on that subject, except to observe that it would be an even more useful and versatile tape deck if it had built-in microphone preamps of some sort rather than just line-level inputs. I did put the DTC-87ES through a series of bench tests, however, and was duly impressed. I was unable to trip it up, not even a little bit, in either the analog or the digital domain. I ran both analog-to-analog and digital-to-analog tests (for the latter I used a digital-to-digital cassette copy of the CBS CD-1 Test Disc), and I can report that the tape deck's performance nudges the limits of the digital medium itself—or at least the limits of my test instruments (see Issue No. 15, p. 47). I saw no nonlinearities, no glitches, no spuria worth mentioning.

As for the control and display features of the DTC-87ES, don't ask what it has, ask what's missing. Because the answer is: very, very little. I would practically have to replicate here the excellent 49-page tabloid-size instruction manual to track through the various recording, playback, and editing facilities. You'll just have to take my word for it—if a consumer DAT deck feature has already been conceived by the human mind, it's almost surely there.

The sound of the DTC-87ES is that of an up-to-date piece of digital audio equipment—transparent, neutral, without a character of its own. (As I said before, if you're looking for lascivious descriptions of upper-midrange liquidity, soundstage width, etc., to inflame your imagination, you're reading the wrong magazine.) I have a few DATs copied from the digital master tapes of forthcoming CD releases (courtesy of producer friends), and they sound exactly the way I expect the CDs to sound. I haven't so far encountered audible dropouts in DAT cassettes, but that doesn't mean they're nonexistent. The most interesting thing that remains to be done is a careful evaluation of the sonic difference, if any, between 48 kHz and 44.1 kHz sampling. That's far from a simple task because the DTC-87ES records analog input signals at the 48 kHz rate, prerecorded digital-to-digital material (CDs and DAT cassettes) at 44.1 kHz, and analog inputs in the long-play mode at 32 kHz, so that only apples-and-oranges comparisons are readily available. "I'll think of something," as the old Brooklyn Dodgers' manager Charlie Dressen used to say to his players before a tough game.

32 " Color TV with Surround Sound

Toshiba CX3288J

Toshiba America Consumer Products, Inc., 82 Totowa Road, Wayne, NJ 07470. Model CX3288J color TV set, $2799.00. Tested sample on loan from manufacturer.

There's little doubt in my mind that the concept of the "home theater" will, in the not too distant future, completely supersede that of the "stereo" (in the sense of "you left your glasses on the stereo, dear"), hence my interest in this somewhat rudimentary, handy-dandy, all-in-one home theater. That interest was further piqued by the set's Carver-designed audio system with "Sonic Holography" and a special "dipole surround speaker." You have to realize, of course, that the whole affair occupies a space less than three feet wide, so that all stereo and surround-sound effects must be launched from within that limited space—quite a feat if it can work at all. Remarkably, it works very well, but don't approach it with high audiophilic expectations.

Here's the configuration. The 32" Toshiba "FST Super Tube" (dual-path electron gun, eight oversized lenses—pretty high-tech) sits at normal couch-viewing level. Directly under it are the left and right main speakers ($2^3/4$" by 5" square drivers in individual enclosures), separated by a center section that houses the remote sensor. Under this center section is the "subwoofer" enclosure with its 6½" driver. The entire video/audio assembly can be swiveled by remote control through an arc of maybe 30 or 40 degrees, and the main speakers can be made to flap backward and forward slightly for stereo adjustments. Superficial bells and whistles like that, along with the thin, black, streamlined, plastic skin in which the whole system is housed, add up to a kind of *Miami Vice* drug-dealer chic.

The dipole surround speaker is actually a pair of opposite-firing 5" drivers in a single detachable enclosure, which can be hung on the back of the set (near the top) or above it. Feeding the L-R signal into a speaker with a figure-eight radiation pattern is actually a clever idea because, when the listener looks into the cusp of the horizontal 8 (i.e., edgewise at the speaker), L-R is completely "decorrelated" from L and R, resulting in the best possible surround effect. The audio power is specified as 10 watts per channel into the main speakers, 20 watts into the subwoofer, and 10 watts into the surround speaker. I don't believe that these specs are subject to the FTC regulations applicable to consumer audio because they appear to be greatly exaggerated. I took no power and distortion measurements, which would have had to be highly invasive, but I did play the opening of the Mahler 5th through the set at a normal, comfortable listening level—and it went crunch. Yes, the little (sub)woofer is tuned quite low, but it can't move much air. Don't make the CX3288J your main audio system.

On the other hand, the MTS/dbx stereo circuitry works very nicely indeed on incoming cable programs, videotapes, and laser videodiscs, with the Carver processor adding some truly dramatic 3-D surround effects. With all four speaker modules in front of you, scrunched together in that small space, you can nonetheless hear all sorts of sounds from behind you and to the far right or left or upstage; in general, all the Indiana-Jones-type extravagant sound effects are fully audible through the CX3288J. That's something of a tour de force, especially with only a phantom center channel. I was impressed.

Here I go, rambling on about audio performance—and this is a TV set. Well, you won't find in-depth video testing in ***The Audio Critic***—not yet—but **I** did run some video tests, using the Reference Recordings *A Video Standard* by Joe Kane, a remarkable laser videodisc that includes, among lots of other things, all the necessary test patterns. I found the black level retention to be good but not great (it never is on consumer equipment); the contrast level obtainable was highly satisfactory long before exceeding the peak linear capability of the set; color performance via the S-Video input was excellent, with the default settings of Color and Tint just about perfect and convergence pretty much beyond reproach at all points. Subjective viewing confirmed the color tests; I can't recall seeing better color on any big-tube consumer set. The geometry could have been better, however; vertical lines tended to be quite noticeably wavy. Toshiba claims 700-line horizontal resolution; that, of course, is irrelevant to real-world video sources (even if it's true—but that's still an area where anarchy prevails); I'm only willing to say that the horizontal resolution of the CX3288J is at least as good as that of the videodisc player I used for the tests, which is specced at 440 lines (with a little exaggeration, I'm inclined to believe). That's still good enough to qualify as state-of-the-art, or close to it, in consumer video.

What's my overall recommendation? Get a separate TV monitor—possibly even a Toshiba FST 32-incher—and a separate audio system, unless you live in a tiny apartment. It should be added that the latest version of this particular Toshiba 32" model is called the **CX3298K.** It appears to be identical in every way, except for its new picture-in-picture (PIP) capability.

Toshiba was also kind enough to send me their top-of-the-line **SV-F990** S-VHS Hi-Fi video cassette recorder to use with the CX3288J. Since the high price of the SV-F990 ($1799.00) is justified only by its stupendous digital effects and editing capabilities, and since I'm not even marginally "into" that video discipline, I decided not to review this VCR rather than to deal with its prime functions inexpertly. I must report, however, that its freeze-frame and slow-motion capabilities are the absolute best in my relatively limited experience, although that's the least of the unit's razzle-dazzle. (I might as well be praising Joe Montana's finger positions on the pigskin.) In terms of ergonomics, I've handled VCRs that I liked better, but none that looked better on the screen.

From now on, I'll try to review at least one video product with a strong audio tie-in in every issue and see where that side road takes us. 0

A Brief Update on CD Players

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

CD players are still the fastest-changing sector of audio. Here are some of the new developments since the article in the last issue.

To begin with, the **Philips CD-80** has been discontinued. Its replacement, the **Marantz CD-72,** will not use the CDM-1 transport but instead the CDM-4 composite transport. The CDM-4 composite replaces many metal parts of the CDM-1 with plastic parts. The SAA7350 one-bit DAC will be used in the CD-72, but the new TDA1547 chip (see the Philips LHH500 review in this issue) will not. Only the **Marantz CD-11 Mk II** CD player will use an alloy-based CD transport and the TDA1547 chip. The Marantz CD-11 will sell for approximately twice the price of the Philips CD-80. The new Marantz CD players will not be available until January 1992. Only **Deltec Precision Audio** (a British firm, no relation to my former company) is currently shipping products with the SAA7350 and the TDA1547. We have been promised a review sample of their **PDM** 2 decoder box. Based on technical data supplied by Deltec, the PDM 2 appears to be a state-of-the-art design. Look for the review in a future issue.

The **Pioneer Elite PD-73** is soon to be discontinued, but the unit is still available as of this writing (though not necessarily by the time you see this print). The PD-73 is the replacement for the PD-71, which I recommended highly in my original article. The PD-73 uses the state-of-the-art Burr-Brown PCM63P-K DAC. The PD-71 used the older Burr-Brown PCM58P-K. The design of the PCM63P-K prevents the low-level linearity problems caused by misadjusted trim pots from occurring. We observed these problems in our sample of the PD-71. The PD-73 continues to include trim pots. These trim pots affect the linearity of large signals only (between 0 dB and -6 dB). Proper adjustment of the trim pots is very complex, and it is unclear if Pioneer is setting the pots correctly. Even with this uncertainty, I highly recommend the PD-73. It is unlikely that we will see another $850.00 CD player with a state-of-the-art power-supply regulator, a quality linear transport, and a top-of-the-line multibit DAC. Based on information found in the service manuals for the two CD players, I can find only one other significant difference between the PD-73 and PD-71. In the PD-73, the digital data decoder has been updated to the Sony CXD1167Q from the Sony CXD1135QZ used in the PD-71. The **Pioneer PD-93** will remain in the line after the PD-73 is discontinued. It has a more robust transport and power supply than the PD-73. In other respects it is almost identical to the PD-73. In the PD-73 all operational amplifiers are used in the inverting mode to prevent common-mode input distortion. This design innovation is not carried over to the PD-93. The price of the PD-93 is $1800.00. If you are in the market for a CD player, run to your Pioneer Elite dealer and purchase the PD-73 before it is discontinued.

The only other high-end design currently using the Burr-Brown PCM63P-K is the **Stax DAC-Talent** ($2700.00), a review sample of which has been promised to us, but the chip is under evaluation at many other well-known manufacturers.

Krell has introduced a new $1850.00 D/A processor called the **Stealth.** Unlike the Theta DS Pro Basic ($2000.00), the Krell uses a standard NPC digital filter chip. The Krell does not use the PCM63P-K but instead uses an inexpensive Burr-Brown PCM67 stereo DAC. The Burr-Brown data sheet states this DAC is "ideal for portable digital audio...ideal for automotive digital audio." A tense and unfriendly dialogue with the Krell staff at the June CES yielded strange hints to the effect that the DAC had been chosen for marketing reasons. (It is a hybrid of one-bit and ladder DAC architecture.) The Burr-Brown data sheets clearly show that the PCM63P-K has lower distortion for low-level signals. If Pioneer can offer a full CD player with the PCM63P-K for $850.00, why does Krell use a cheaper chip in the Stealth processor?

PS Audio has changed the enclosure for the Digital Link from a modem box to a standard full-sized equipment enclosure. This addresses the principal complaint ***The** Audio Critic* had with the original Digital Link. The price for the new unit is unchanged; thus the Digital Link represents an even better bargain. A complete report on this new **Digital Link Series II** will appear in the next issue.

Aragon has also modified the D2A digital decoder box. The new unit is called the **Mark II D2A.** Two of the modifications address problems we found in the original design. The output is now buffered with a source follower, and the differential pairs are now biased by current sources. In addition, the digital section has been modified to further reduce jitter in the recovered clock. The price of the D2A has unfortunately increased to $1600. Of that, $300 is to cover cost increases incurred in the manufacture of the D2A. I never did understand how the unit could be produced at just under $1000. The additional $300 of the price increase is for the larger IPS external power supply. The IPS supply, which was an option, is now shipped with all units. The $300 price increase can be justified by the increased parts cost for the IPS over the standard supply. The larger supply is claimed to improve the sound quality of the unit. I could not detect any sonic difference between the D2A when driven by the standard power supply and when driven by the IPS. This comes as no surprise, since the standard power supply was more than adequate to drive the D2A. At $1300 the Aragon Mark II D2A would be a great deal. In comparison with the PS Audio Digital Link, it has lower clock jitter, a more complex digital filter, and significantly reduced RF output level. At $1600, the Mark II D2A becomes a bit pricey as the price approaches that of the Theta DS Pro Basic ($2000). Perhaps Aragon can be persuaded to ship the original power supply again. I will have a complete report on the Mark II D2A in the next issue of ***The Audio Critic.***

Enlightened Audio Designs is about to enter the D/A processor race. The unit will sell for $1299.00. It will use the NPC SM5813 digital filter, the Analog Devices AD1862N-J DAC, and the EAD proprietary current-to-voltage converter. (See also the EAD review in this issue.) The S/PDIF decoder is still under development at this writing. EAD has been very secretive about the design of the current-to-voltage converter because patents are still pending. From the little information I was able to pry from the company, it appears that the design approach is valid and innovative. One detail of the design that was revealed was that voltage feedback is used instead of a transimpedance amplifier. EAD engineers argue that the first stage of the reconstruction filter cannot in most cases be incorporated into the current-to-voltage converter when a transimpedance amplifier is used. This problem occurs because most transimpedance amplifiers will oscillate when a capacitor is placed in the feedback loop. If the first stage of the reconstruction filter is not incorporated into the current-to-voltage converter, then the settling requirements for both the current-to-voltage converter and the filter stage become more stringent. Other manufactures, such as MSB Technology, have overcome the problems of using transimpedance amplifiers, but the EAD argument is valid.

Wadia has just introduced an analog-to-digital processor. This device is for use with a DAT recorder or professional tape recorders. The UltraAnalog analog-to-digital converter is used in this unit. The UltraAnalog ADC uses a brick-wall analog antialiasing filter and not the Wadia time-domain algorithm. It will be interesting to see the impulse and square wave response of the complete Wadia system using both the new analog-to-digital processor and a Wadia decoder box.

Some high-end manufacturers (Audio Research, Barclay Audio, and Wadia) have recently suggested that the S/PDIF coaxial and optical interfaces are not adequate. They have proposed a new optical interface using the AT&T ST-type glass fiber input and output connectors. Apparently these companies are attempting to have a standard established by the Academy for the Advancement of High End Audio (AAHEA). You may wonder why the proposal is not being made to the normal standard-setting committees in audio such as the AES or IHF. The reason is that the change to this very expensive AT&T interface cannot be justified on scientific grounds. *[The AAHEA is a chamber-of-commerce type of group without any scientific credentials. The word "Academy" in its name is a joke.—Ed.]* The AT&T data link was designed for very long cable runs and not a 2-foot run between a CD player and a decoder box. The claim that the AT&T link "sounds better" would never be accepted by a professional standards community.

Many major audio designers have privately expressed the opinion that the AT&T link is a waste of money. These manufactures are afraid that they will have to incorporate the AT&T link if the standard is established by the AAHEA This will result in a major price rise to cover the cost of the AT&T link. If high-end companies want to establish a better data transmission standard for the link between a CD player and a decoder box, they should should consider the method used in the Sony DAS-R1 (see my article in Issue No. 15). Sony connected two data lines between the CD player and the decoder box. The second data line carried a low-jitter data clock, generated in the decoder box, to the CD player. This data transmission method would significantly reduce jitter in the clock connected to the digital-to-analog converter. It could also be incorporated with only a very small cost impact.

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

By Peter Aczel
Editor and Publisher

Here we come to the technical examination of the subject, as announced in the last issue. This part deals with the amplifier/speaker interface and the effects of wires/cables at that junction in the system.

I wouldn't be entirely forthright if I didn't state right up front that this article is, in a sense, quite unnecessary. In the August 1989 issue of *Audio,* Richard A. Greiner, Ph.D., professor of electrical and computer engineering at the University of Wisconsin, published an article under the title of "Cables and the Amp/Speaker Interface," which in turn was an updated adaptation of his original paper, "Amplifier-Loudspeaker Interfacing," published in the May 1980 issue of the *Journal of the Audio Engineering Society* and presented a year earlier at the AES convention in Los Angeles. Everything of substance I'm about to say on the subject of speaker cables has already been explained—100% correctly, lucidly, and in great detail—by Professor Greiner; I can only add my own little flourishes, commentary, and illustrations. For some perverse reason, the rank and file of audio consumers will give credence to the most ignorant exudations of gonzo audio journalists and loudmouthed dealers while tending to regard with suspicion and skepticism a superbly accredited and commercially disinterested authority like Dick Greiner. I was disgusted by some of the reactions to the *Audio* article, and I offer what follows here in the faint hope that I can tip the scales back—even if only part of the way—to sanity.

(By the way, as some readers may still remember, I had a little tiff with the professor a good many years ago, in my "Letters to the Editor" column. I overreacted in a needlessly intemperate manner to a mild bit of professorial pomposity, which at the time I perceived as condescension, and he took offense. Actually, I have the greatest respect for the man and wish in retrospect that the contretemps had never taken place.)

What cable cultists never think about.

For openers, let's face a few simple facts of life. Such as:

Inside a large and complicated loudspeaker system there may be as much wire, or more wire, than between the amplifier and the speaker terminals. It starts with the voice coils (a single turn of one those 4-inch JBL voice coils is over a foot long—and how many of those turns are there?) and continues with all the wires connecting the individual drivers to the crossover network, the wiring inside the crossover network (including large coils), and then the wiring from the crossover to the outside terminals. Or take the Quad ESL-63, a particularly poignant example, with the staggering length of thin, nontweako wire in its unique delay line. Then, of course, there's also a significant length of wiring inside the amplifier before the output is brought out to the terminals. In the case of tube amplifiers, add to that the great length of wire in the output transformer. The cable cultist has absolutely no control over the dimensions, geometry, or metallurgy of these hidden wires and cables—even if such dimensions, geometry, or metallurgy were of serious sonic importance. It's like being a health-food faddist at lunch but not at breakfast or dinner. Thus, before any discussion of engineering considerations, irrationality raises its bony head. (Or did you think Celestion wires the inside of the SL700 speaker with MIT Music Hose?)

Another fact that needs to be faced from the start is that music, or any other audible program material, consists of frequencies from about 15 or 16 Hz to 21 or 22 kHz. (I'm being very generous and therefore assume state-of-the-art recording and 16-year old hearing prodigies.) Let's expand that bandwidth to 50 kHz, however, since it doesn't cost us anything in an abstract argument and will make bandwidth fetishists happier. Surely, no information above 50 kHz needs to be transmitted by the amplifier to the speaker. Is a speaker cable's performance above 50 kHz relevant then? Does it have to be a good microwave transmission cable? You know the answer, but keep it in mind as we examine the network characteristics of speaker cables.

Let's also agree, before we proceed, that a direct connection from the amplifier output terminals to the speaker input terminals—perhaps with an inch or two of bus bar or braid but without any cable as such—is the theoretical ideal and that nothing can be more accurate than that. Ask a cable cultist what's better than pure silver cable, or any other cable, and he'll be most unlikely to give you the obvious answer, which is also the scientifically correct answer: no cable at all. That will be our standard of perfection for the purposes of this study.

Modeling the amplifier/speaker interface.

This is where my small contribution comes in—my doodles, as it were, on the margins of the Greiner articles. I claim absolutely no originality here; all of what I'm doing is quite straightforward and ordinary; however, I haven't so far seen the real-world effects of speaker cables illustrated in exactly this manner anywhere else.

As Dr. Greiner points out, the amplifier/cable/speaker interface can be represented by the lumped-element equivalent circuit shown in Figure 1. This is a sufficiently accurate representation for our purposes; treating the cable as a transmission line is theoretically "purer" but a total waste of time, considering even the longest cable runs and highest frequencies encountered in audio work. (Did I say 50 kHz? That's a wavelength of 6 kilometers!) Thus, a length of cable between

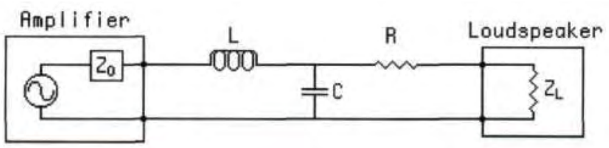


Figure 1: Equivalent circuit for the amplifier/cable/loudspeaker interface, with lumped circuit elements.

the amplifier and the speaker is, electrically speaking, a series inductance, a shunt capacitance, and a series resistance. That's all it is, really, unless you get involved in second-order and third-order effects that have no influence on the transmission of audio frequencies over domestic distances, e.g., skin effect, which is also called radio-frequency resistance (although the high-end audio cable touts would rather die than refer to it by that self-stultifying name). Once you have characterized a speaker cable as an RLC circuit, you can predict with considerable precision its effect on the network which it forms with the source (viz., the amplifier) and the load (viz., the loudspeaker).

Luckily for me, Martin Colloms (the noted Jekyll-and-Hyde audio journalist in England, who does excellent technical work but talks audio-salon voodoo) has already measured the RLC values of 44 name-brand speaker cables, thus sparing me the trouble of doing the same. He published the results in the July 1990 issue of *Hi-Fi News & Record Review,* and I trust his figures as completely as I am dumbfounded by his grading of the "pace," "ambience," etc., of each cable. (I have a fork that brings out the piquancy of sauerbraten like no other, Martin.) I can now plug the Colloms data into a circuit analysis program on my computer and obtain the response curve of any network formed by a known amplifier, one of the 44 cables, and a known loudspeaker system. Such a response curve will be accurate to the extent that the source and the load are modeled accurately.

The program I use is a relatively simple one: Micro-Cap II Macintosh Professional Circuit Analysis Program, Version 2.71, by Spectrum Software of Sunnyvale, California. The amplifier I used for modeling the interface in most of the analyses here was my trusty Boulder 500AE, which can be represented as a source impedance by an R of 0.01 ohm in series with an L of 2 µH—almost a perfect voltage source. I also did a few runs using the much more current-sourcey Carver Silver Seven tube amplifier instead, modeled by an R of 1.1 ohms. These values derive from actual measurements. The speaker system I chose to represent the load in my network model was the Carver "Amazing Loudspeaker" Platinum Mark IV, not so much because it's one of my favorites but because I was able to obtain a very accurate circuit diagram of it, showing every crossover and equalization component value plus the equivalent circuits of the transducers, including the motional impedance of the woofer system. I've decided not to reproduce the schematic here because I want to keep this discussion focused on speaker cables, not an interesting speaker design; just take my word for it that we have a nice, fairly complex, real-world load here, but not so difficult to drive that it could be objected to as untypical.

What the simulated response curves show.

Let's start with the aforesaid ideal situation, where the loudspeaker is being driven from an almost perfect voltage source (viz., the Boulder) without any cable—amplifier output terminals into speaker input terminals. Figure 2 shows the frequency response at that junction and proves that the fancy load represented by the Carver speaker looks barely different from a resistor to a voltage source. (Note that the upper limit of these simulations is 100 kHz—to forestall bandwidth arguments, as I've said—but it so happens that the Boulder does have a small-signal bandwidth of 200 kHz.)

Now let's insert 10-meter lengths of various speaker cables between the amplifier and the speaker to see how their different RLC values affect the response at the speaker input terminals. In a fair-sized room where the equipment, including the amplifier, is at one end and the speakers are at the other, 10 meters (32.8 feet) is a typical cable length, especially if the cable is routed along the baseboard or otherwise not dressed in a straight line.

Figure 3 shows the response with the least inductive and most capacitive cable modeled here, the AudioQuest Clear Hyperlitz ($50.00 per foot, plus $95/pair for prep). The low inductance limits the lowpass filter effect, but the 0.4 dB drop from 7 kHz to 12 kHz may conceivably be audible to the critical ear. I also want to mention that the MSSigma Series by Monster Cable (almost as costly) has highly similar RLC characteristics and will yield a virtually identical response.

Taking the cables in their order of increasing inductance and decreasing capacitance, we come to the Kimber 4AG braided silver cable, at $100 per foot (welcome to cuckoo country). Figure 4 shows the response. With about 50% higher inductance, 65% higher resistance, and totally

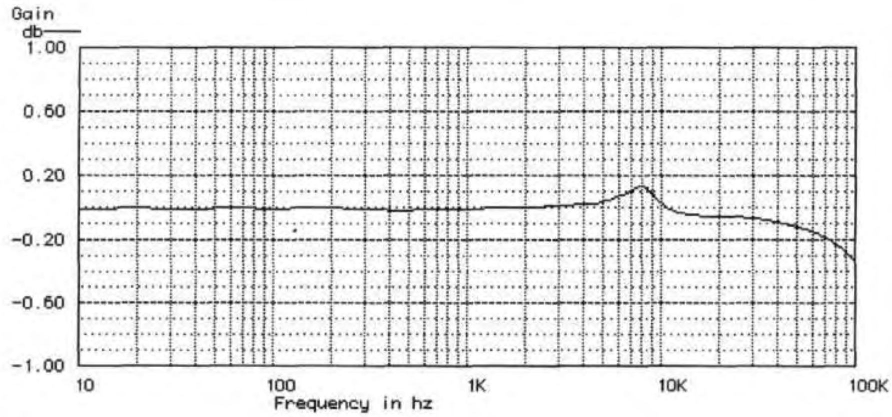


Figure 2: Response at the speaker input terminals with direct feed (no cable) from the Boulder amplifier. Note that the response stays flat within ±0.13 dB from 10 Hz to 50 kHz.

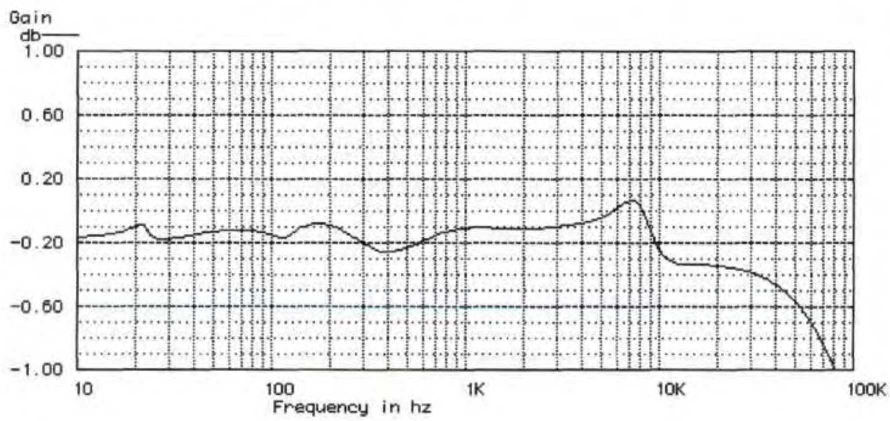


Figure 3: Response at the speaker input terminals with 10 meters of Audio-Quest Clear Hyperlitz cable driven from the Boulder amplifier. Note 0.4 dB drop from 7 kHz to 12 kHz.

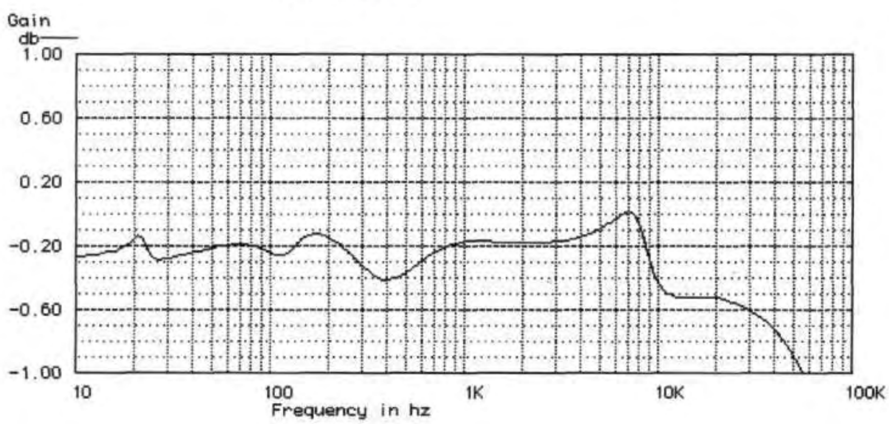


Figure 4: Response at the speaker input terminals with 10 meters of Kimber 4AG cable driven from the Boulder amplifier. Note 0.5 dB drop from 7 kHz to 11 kHz and 400 Hz notch.

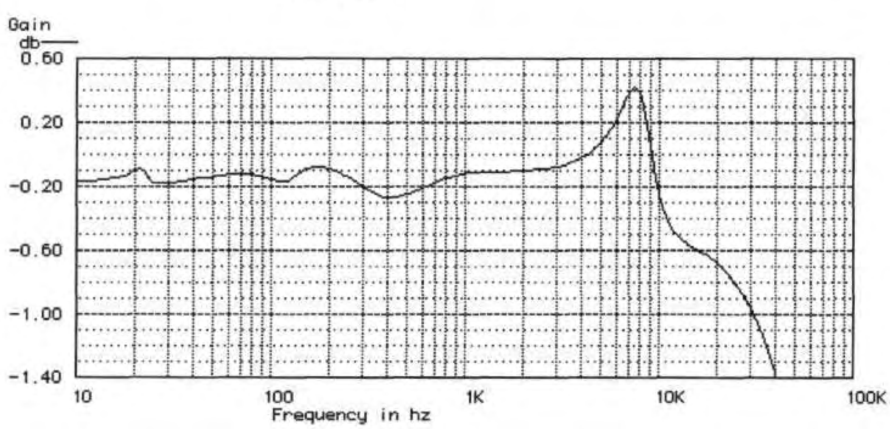


Figure 5: Response at the speaker input terminals with 10 meters of Monster Cable Standard driven from the Boulder amplifier. The drop from 7 kHz to 20 kHz is 1.1 dB.

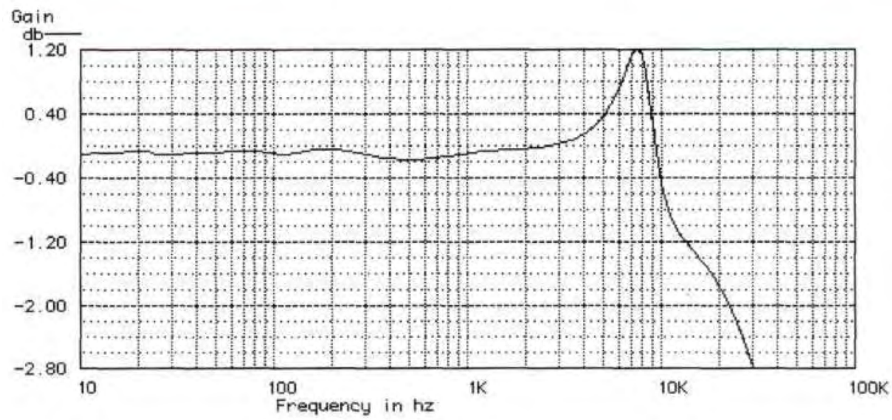


Figure 6: Response at the speaker input terminals with 10 meters of Siltech Ribbon cable driven from the Boulder amplifier. The drop from 7 kHz to 20 kHz is 3 dB. Note scale change.

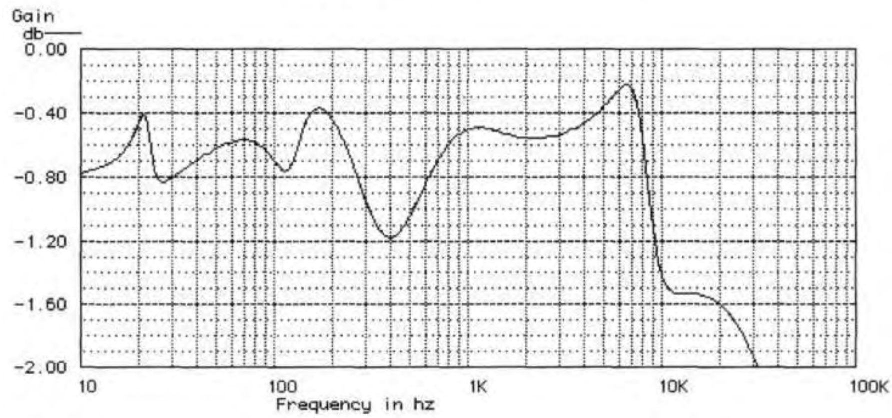


Figure 7: Response at the speaker input terminals with 10 meters of Vecteur 0.8mm solid copper cable driven from the Boulder amplifier. Note scale change back to original.

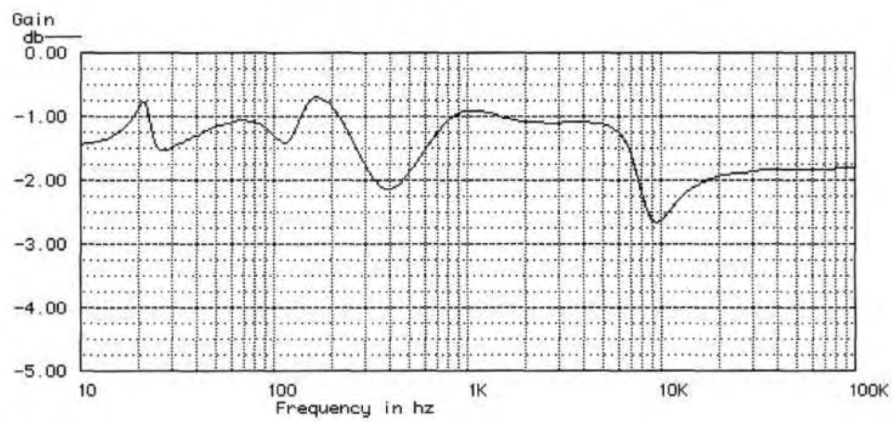


Figure 8: Response at the speaker input terminals with direct feed (no cable) from the Carver Silver Seven vacuum-tube amplifier. Note new scale change to coarser divisions.

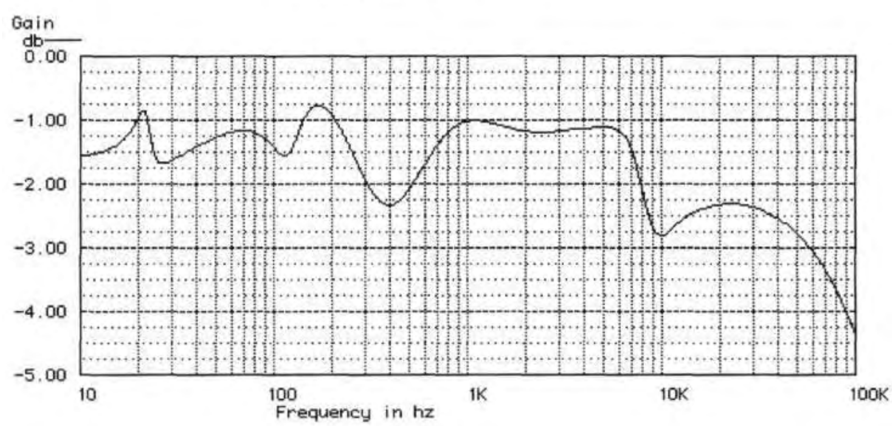


Figure 9: Response at the speaker input terminals with 10 meters of Monster Cable Standard driven from the Carver Silver Seven vacuum-tube amplifier. No change from direct feed!

different metallurgy/geometry, everything is worse by about 0.1 dB, without a change in overall profile. Big deal.

The relatively cheap Monster Cable Standard is next in line. It's almost four times as inductive as the AudioQuest and the response, as shown in Figure 5, is beginning to look like that of a mild lowpass filter. If a critical listener reported a slight softening of the top end with this cable, I wouldn't be the least bit surprised. Up to 3 kHz, however, the response is identical to that of the AudioQuest. Same bass, same midrange—not much possibility of an audible difference there.

Shall we go to extremes? Let's try a crazily inductive cable like the Siltech Ribbon from the Netherlands, by far the costliest of them all, made of extruded silver ribbon with perfect crystal structure, etc., etc. At approximately 2 µH per meter, it throws caution to the wind inductancewise, and a 10-meter length gives the response shown in Figure 6. Now that's a lowpass filter that even tin ears will easily hear in this particular system. (Martin Colloms heard it, too, and wrote, "Head and shoulders above the rest [the other 43 cables] was the Siltech Ribbon; yes—one hell of a price, but what accuracy!" Now, Martin used only a 5-meter length of cable, so he was putting 10 µH between his amplifier and his "predominantly...resistive 4-ohm" speaker, the KEF 105/3. A rough calculation translates that to a 2.4 dB droop at 20 kHz. That's accurate? Maybe to a golden ear...) This is clearly not the cable for long runs, unless the impedance of your speaker rises dramatically at the higher frequencies (and your banker calls you Mr. Getty).

Figure 7 illustrates a special case, that of the Vecteur 0.8mm solid copper cable, basically a tweako cult item but carrying a guarded endorsement by the illustrious Dr. Malcolm Hawksford *(Hi-Fi News & Record Review,* August 1985—and don't ask me to explain what he means). This is a much more resistive cable than the others; the 10-meter length modeled here represents a series R of 0.56 ohms, and its inductance is also quite high, between that of the standard Monster Cable and the Siltech Ribbon. The result is a weird roller-coaster-plus-lowpass-filter profile, not very promising sonically, unless you think an undulating ±0.7 dB response across the audio range is more acceptable in a speaker cable than in an amplifier.

But you ain't seen nothin' yet, folks. Take a look at Figure 8. That's a direct-feed, no-cable situation just as in Figure 2, except that the amplifier is the Carver Silver Seven, with its 1.1 ohm output impedance. It isn't only wire in the signal path that can alter the response! Here we have a ±1 dB characteristic, with most of the energy below 7 kHz on the plus side and everything above 7 kHz on the minus side. No wonder audiophiles talk about the "tube sound." A 2 dB range of fluctuation across the spectrum can be expected to be audible.

Here comes the mindblower. Figure 9 shows what happens when the Monster Cable of Figure 5 is used with the Carver Silver Seven instead of the Boulder. Nothing happens! The high-output-impedance signature of the tube amplifier is so dominant that up to 20 kHz the response is the same as it would be without the cable—and we're talking about a cable that has a distinct lowpass filter effect on this system when driven from a voltage source. Your typical high-end reviewer would probably report that the Carver amplifier isn't at all cable-sensitive—or maybe that Monster Cable Standard is somewhat amplifier-sensitive. "Where ignorance is bliss, 'tis folly to be wise," says the poet.

What does it all add up to?

The conclusions to be drawn from the above are fairly obvious, but let's spell them out.

No speaker cable of significant length is "accurate" in the sense that the signal is the same, or virtually the same, at the speaker end as at the amplifier end, but those with lower series inductance are more accurate than those with higher series inductance, as long as the series resistance is reasonably low. Metallurgy is irrelevant to accuracy, and construction is relevant only to the extent that it controls the series inductance per unit length (and, possibly, the cable's susceptibility to RFI, a subject I have yet to address). Price is also irrelevant, except that very low-inductance speaker cable is never dirt-cheap. Shunt capacitance is of little or no consequence as long as the amplifier is perfectly stable, an assumption made in all of these simulations but not always the case in the real world. Finally, if the amplifier isn't a voltage source—i.e., if it has a high output impedance—all cable characteristics will be swamped, except in the most extreme cases.

What about the sound? Obviously, two speaker cables as similar in response as, for example, the AudioQuest Clear Hyperlitz and the Kimber 4AG can be expected to be indistinguishable in a double-blind listening test. As I have always insisted, A and B will inevitably sound the same unless there exists some kind of mechanism whereby they can sound different. (Weird reasoning, isn't it?) In this case, a difference of 0.1 dB is an insufficient mechanism. On the other hand, a cable like the Siltech Ribbon is so different in response from the others that I'd be astonished if an experienced audiophile couldn't distinguish it by its sound. The point is that speaker cables will sound the same or different according to their RLC characteristics, not according to the voodoo criteria of the cable cultists. Thus, if you inserted a small circuit board with the proper RLC values—costing maybe $2.00 or thereabouts—between the amplifier and the speaker in the direct-feed signal path of Figure 2, you could obtain the Kimber 4AG silver cable's exact response as shown in Figure 4, at a saving of thousands and thousands of dollars. (That's Larry Archibald's and Dick Olsher's cable, if you'll forgive me some name-dropping.)

So what's the best thing to do?

The best advice must be practically staring you in the face at this point. Simply avoid long runs of speaker cable—any speaker cable, no matter how good you think it is. In most installations, that's eminently doable. With a pair of mono amplifiers, you place each amp directly behind each speaker and make the connection with a minimum amount of wire—any kind of wire. When you're talking inches or a foot, the RLC values simply don't matter. Or, if you have a stereo amplifier, place it right between the two speakers and use four of five feet of wire to connect each speaker. Make it 16-gauge or thicker—ordinary lamp cord is fine—and forget about the L and C values because they'll be quite negligible at that length. The whole thing becomes a nonissue.

Where do you put your preamplifier? With balanced lines, you can put it at any distance from the power amplifiers). With unbalanced lines, you can usually put it just as far away, but make sure that you have no hum and no RFI. In the worst case, if you have serious problems with long unbalanced lines, put all your stereo components between the speakers, especially if you play mostly CDs. (Only turntables tend to be affected by the sound field in the proximity of the speakers.) In the age of the remote control, such a deployment—with short wiring everywhere—has become quite convenient. Use long speaker cables only as a last resort. What kind, if you must? Chris Russell, mastermind of the Bryston amplifier company, recommends RG-8 coaxial cable, which is lower in inductance than spaced 2-conductor types and only slightly higher in capacitance (meaning that the 10-meter profile would fall somewhere between Figures 4 and 5), has a 13-gauge center conductor, and costs 42 cents per foot at Radio Shack. Now *that* sounds good to me.

One more thing.

Before I sign off—until Part III, that is—I'd like to return very briefly to the bandwidth issue and register a word of protest against what I consider to be the most misleading speaker cable advertising of all—because it looks so scientific on the surface. I'm talking about those highly technical MIT (Music Interface Technologies—definitely not Massachusetts Institute of Technology) ads and brochures showing all kinds of oscilloscope pictures of impulse response, "phase noise" (their term, not mine), and other time-domain performance characteristics of MIT cables, in documentation of their alleged technical superiority. The trouble is that the time axis in the scope pictures either isn't labeled at all, or else the time-per-division information is buried somewhere in the small print. The technically unsophisticated audiophile looking at the ads and brochures is under the impression that he is being shown superior performance in the audio range, whereas in reality all of that time-domain action is happening in nanoseconds, totally unrelated to the audio range (which extends, even with our agreed-on stretching, only from 67 milliseconds to 20 microseconds). MIT is selling *megahertz* performance to the *audio* market for big bucks. Not that they're the only snake-oil artists among the cable vendors, but I happen to be particularly irritated by their kind of scientific non sequitur. The only thing that irritates me even more is that a few years ago I allowed one of those ads to slip through into the pages of this publication. I don't think, however, that cable advertisers will be breaking down my door from now on. ¥

Hip Boots

Wading through the Mire of Misinformation in the Audio Press

Editor's Note: *David Rich, our Contributing Technical Editor, whose opinion I obviously respect, keeps telling me to stop assailing the "loony tunes" of audio in this column and address only the big, complex untruths, such as, for example, the "audible" superiority of $6000 amplifiers to $1200 amplifiers. He feels that once the major misconceptions are dispelled, the loony tunes will wither away. I'm not convinced. If the subject is big enough, I certainly want to run a full-length article about it, but if some loony starts writing about, say, a magic aerosol spray that makes your cables sound better, I still don't want a music-loving real-estate agent from Atlanta to think that it just might be true. That kind of irresponsible drivel deserves instant ridicule, and that's where this column comes in. And, of course, if the loony should then decide to attack me and this publication.... So "Hip Boots" goes on.*

The endemic idiocy of the tweako/loony subculture within the audio community is the desire to improve whatever needs no improvement—in other words, to put time, energy, and money into solving already solved problems, or nonproblems—while paying no attention to the remaining weak links in the chain of sound reproduction. Thus the power from the wall outlet must be made purer, the zeros and ones in a digital storage medium must somehow be made more zeroish and oneish, copper wire must be made a better carrier of electrons, and so forth—but a pair of bookshelf speakers with a bass cutoff of 80 Hz, ±5 dB frequency response, and no power-handling capability will do just fine to verify all of the above. *O sancta simplicitas!*

David Zigas and Tim Smart in *Business Week*

Business Week is McGraw-Hill's prestigious and ostensibly authoritative weekly magazine on business and financial matters. David Zigas is listed on the masthead as Associate Editor for Corporate Finance; Tim Smart is listed as a Washington correspondent; obviously both report to higher-echelon editors. I'm reasonably certain that neither of these professional business journalists would file a story about a new development in, say, the steel industry without obtaining corroboration from a number of highly reliable sources. It seems, however, that audio isn't important enough at *Business Week* for that kind of accountability.

In the "Personal Business" section of the June 25, 1990 issue, "Stereo" subdivision, David Zigas comes out as an unabashed shill on behalf of the $1250 Tice Power Block, the utter nonsensicalness of which is discussed in a separate commentary below. Try it, you'll like it, says McGraw-Hill's trusted editor, without betraying the slightest knowledge of the contempt in which the device is held by the scientific audio community. Tim Smart is no smarter in the January 28, 1991 issue, again in the "Personal Business" pages, under "Music." He goes gaga over "high-purity" cables, $100 silver extension cords, CD rings and clamps, CD edge treatments and sprays—the whole tweako toyshop. He writes that green paint for your CDs and those slithery/tacky isolation feet for your CD player are "already in the mainstream." (The feet, by the way, "help absorb excess electrical energy." How can such untutored technobabble get through the editorial process at *Business Week!)*

Well, I've got news for you, Tim. The mainstream of audio is represented by the AES , the IEEE, and the ASA, not by the pimply-faced "consultant" in a Bon Jovi T-shirt at your local audio salon. And that educated mainstream says: don't just assert that you can hear these "improvements," for which there are no genuine scientific rationales, but prove in a controlled double-blind test that you really can. Because the mainstream practitioners, and I, can't.

Anent George Tice in *The Absolute Sound* et al.

Talk about hip boots. You definitely need to put on a pair when you tread where George R. Tice of Tice Audio Products has left a wake—which is of course all over the high-end audio press. There's the review of the Tice Power Block "line conditioner" and Titan "energy storage system" in the January/February 1990 issue of *The Absolute Sound* and the April 1990 issue of *Stereophile,* among others. There's the unbelievable endorsement of the Tice TPT Clock in the November/December 1990 issue of *The Absolute Sound,* and then there's all the grotesque literature that Tice has put out on these subjects—and that's not all, but I can't be bothered to track down every loony tune in the business. To respond to all these outrageous claims tit for tat with scientific arguments would take far too many pages (for which I have better uses, as this issue demonstrates), but I feel the need to go on record thusly:

The Power Block is garden-variety high-end audio nonsense and a more or less typical rip-off, whereas the more recent TPT Clock is an insult to our intelligence and the last straw, which we might have been spared had the tweaks and cultist not emboldened Tice in the course of his previous endeavors. At this point he appears to be utterly shameless.

Why does the Power Block fall into the category of taurine excreta? First of all, because the overwhelming majority of audiophiles will never need a power line conditioner. Secondly, because the ones who may conceivably need one are those with relatively inexpensive equipment (no regulated power supplies, inadequate rejection of high-frequency power-line artifacts, funny grounds, etc.), whereas those who can afford a $1250 Power Block will most probably be using carefully engineered high-end electronics with all sorts of built-in features and safeguards that make the Power Block redundant and possibly even counterproductive. Thirdly, because for approximately one-third the price of the Power Block you can buy a more capable high-current power line conditioner from a computer supply house. Fourthly, because in those rare cases when a power line conditioner is actually helping you, the benefit is *not* in terms of imaging/soundstaging/liquidity and the rest of the tweako reviewers' fetish kit. More likely, you don't hear the fridge turning on anymore and the good buddies of the CB crowd no longer interrupt your Mozart with their ten-fours—that sort of thing. Fifthly...never mind. I said this wasn't a detailed technical analysis, and besides there are other overpriced line conditioners in the audio marketplace now that have no greater credibility. But Tice originated the genre and then set a new record in lack of credibility with the TPT Clock.

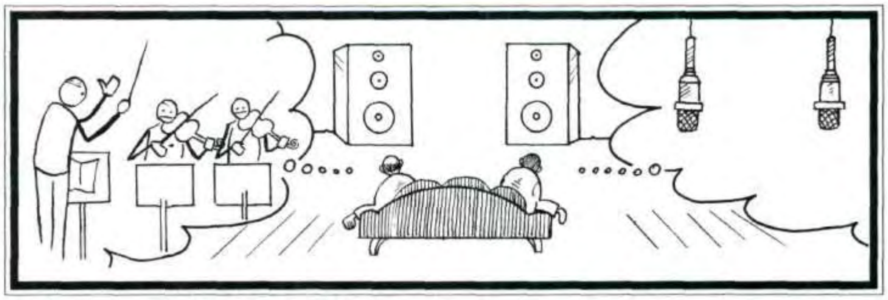
The trouble is, I don't really want to talk about the Tice TPT Clock. Some claims—just a very few—are so stupid and so insincere that by taking them seriously enough to refute them one gives them an undeserved measure of temporary credibility. The claim that by plugging a specially "treated" (blessed?) digital clock into the wall you will obtain "corrected" electron flow for your audio components —with immense sonic benefits, of course—is such a loathsome piece of charlatanry that I refuse to say anything more about it than this:

If George Tice can produce three electronics experts with university graduate degrees in engineering or physics who are not commercially linked to him and who will certify in writing that his claims for the TPT Clock are scientifically valid, then I shall devote a special issue of *The Audio Critic* exclusively to the explanation and celebration of his technology and mail it as a free bonus to all subscribers. Fair enough? 0

An open mind is all very well in its way, but it ought not to be so open that there is no keeping anything in or out of it. It should be capable of shutting its doors sometimes', or it may be found a little drafty.

—SAMUEL BUTLER (1835-1902)

Recorded Music



The heading of this column has been changed from "Records&Recording" to "Recorded Music." Records mean LPs, not CDs (when someone mentions a shelfful of records, you don't think of a row of little see-through plastic boxes), and this is now strictly a CD column (DAT-ready, to be sure). The CD has clearly superseded the LP; in fact, significant new recordings available only in the LP format are nonexistent. Virtually all new recordings are digital, and converting digital information into analog vinyl grooves makes very little audio sense. Yes, there remains a minuscule contingent of die-hard analog recordists, but their analog master tapes also end up being archived to CD as well as LP, and the CD version generally sounds better—or at least cleaner, without the ticks, pops, and swishes.

Mehta and the New York Philharmonic to the Max (Wilcox, That Is)

By Peter Aczel
Editor and Publisher

The best recording engineers are *almost* producers, and the best producers are *almost* recording engineers. Max Wilcox falls into the latter category; indeed, he goes a step beyond it, as he is quite capable of making a recording all by himself, although he likes to have a first-rate technical team at his elbow. Overall, however, he is very much the musical producer and not the techie, in contrast to a John Eargle for example, who is primarily an audio expert with a strong background in music. John teams up with Adam Stern as his producer partner at Delos; Jack Renner relies on Robert Woods at Telarc; but Max Wilcox relies almost entirely on himself for achieving the desired artistic/sonic result in recording and has worked with many different engineers for many different labels. His basic perspective is always that of a musician (he is a pianist, piano coach, and sometime conductor), and for that reason his recordings, whatever their specific audio qualities may be, are always balanced and musical, never eccentric or "experimental" in sound. Max has taste and common sense.

Now that I've known Max off and on for something like twenty years (old-time subscribers will recall the series of articles he contributed to early issues of this estimable journal), I retrospectively discern a steady improvement over that period in the transparency and resolution of his recorded sound, whereas the basic correctness of his balances, emphases, and spatial relations has never changed. I wish to take a little bit of credit for the improvement, having been one of proselytizers who converted him to state-of-the-art hardware, fewer mikes and tracks, and minimal signal paths. Before then he was, hardware and multitrackwise, a *bien-pensant* RCA company man of the early 1970s.

Today Max is clearly in the same league with the cult names usually linked to audiophile-quality recorded sound, but he parts company with that crowd in at least one significant respect. Just about all of his recording experience has been with world-class musicians—Rubinstein, Ormandy, Solti, the Guarneri Quartet, Peter Serkin, Richard Goode, and others in that outstanding-to-great bracket—so that he necessarily has higher expectations as he records and edits than a recordist accustomed to lesser forces. That can't help but affect the audio quality of the effort.

New York/Mehta with Wilcox on Teldec

The New York Philharmonic, long associated with CBS Masterworks, now records on the Teldec label. When Teldec's A&R director, Wolfgang Mohr, launched Zubin Mehta's swan-song series of recordings with the New York orchestra (1990-91 is his last season there), Max Wilcox got the job of producing the sessions. The venue chosen for the recordings—by a process of elimination and not without misgivings, I understand—was the old Manhattan Center on 34th Street, now a union meeting hall and not exactly a recordist's dream. It requires some fancy acoustical footwork to make a half decent symphonic recording there. The quality of orchestral sound obtainable from these three CDs on good playback equipment is therefore a near miracle and a testimonial to Max's skill and perseverance. This is his finest work to date; the orchestral textures, balances, and dynamics are at least as good as, and possibly better than, I've ever heard on any label; the spacious, expansive acoustic is generic, to be sure, rather than hall-specific—meaning that Manhattan Center sounds like an unidentifiable *good* hall—but even the decay characteristics are lovely and convincing despite the light sprinkling of artificial reverb I know is there (although I don't perceive it as such). I particularly like that the spectral center of gravity, so to speak, is in the lower midrange, as it is in real life, rather than slanted toward the highs as in so many hi-fi spectaculars. This is a rich, luxurious sound with the correct proportion of clear, delicate highs, tremendous inner detail, and the authentic weight of a full symphony orchestra. The mikes used were all Sennheiser and Schoeps omnis—and I must take back all my previous negative remarks about Sennheisers.

The playing of the orchestra has, of course, a great deal to do with the sonic impact of the recordings. The New York Philharmonic is well known for the lackadaisical or even goonish reponse of its virtuosi to certain conductors under certain circumstances, but here they give their valedictory maestro all they've got and truly sound like "one of the five greatest orchestras in the world" (Mehta's assessment as quoted in *Fanfare* magazine). There's a difference between this level of playing and that of, say, the Seattle or Atlanta orchestras, excellent as they are.

My comments on the individual CDs follow, in the chronological order of the recordings, which took place between September 1989 and January 1990. The technical team and digital recording equipment were supplied by New York Digital Recording, Inc.

Mahler

Gustav Mahler: Symphony No. 5 in C-sharp Minor. New York Philharmonic, Zubin Mehta, conductor. ***Teldec*** *2292-46152-2 (DDD, recorded September 1989, released 1990).*

This is almost surely the greatest music in the series and perhaps also the best-sounding recording, although it was mixed down from multitrack, whereas the others were done live to two-track. It may be that the humidity was just right because the central heating had not yet been turned on. It's a subtle difference, in any case.

The playing here is virtuosic beyond belief; in fact, such a performance is most unlikely to have taken place in real life but must be the result of very skillful editing. No matter; at least we know that the conductor and the producer cared enough to spare no effort, and the end result is what counts. (I'm not of the school of "hey, you cheated me because I can't hear the mistakes.")

As for Mehta's interpretation, he doesn't have a Bruno Walter's or a Jascha Horenstein's emotional or stylistic commitment to Mahler, but he is a superb musician with a baton technique all the way up there in the Reiner class, and the resulting precision, clarity, shapeliness of phrasing, and beauty of sound serve Mahler very well indeed. Unless you insist on the utmost Austro-Bohemian *Weltschmerz* and *Galgenhumor* in your Mahler—or a lot more hysteria a la Bernstein—this is a very good 5th to own.

Hoist

Gustav Hoist: The Planets, Op. 32. New York Philharmonic, Zubin Mehta, conductor. ***Teldec*** *2292-46316-2 (DDD, recorded November 1989, released 1990).*

Much the same observations apply here regarding the interpretation as under the Mahler heading above. Mehta somehow streamlines and internationalizes the roast-beef-and-Yorkshire-pudding Britishness of the score; this is not in the juicy, easygoing idiom I remember from Sir Adrian Boult. I wish the big tune in the Jupiter movement were played more like a second "God Save the Queen." (But why should an Indian born in preindependence Bombay want to think British?) Even so, in a hard-driven, brilliant, bravura performance, Mehta makes a positive impression with his clarity, precision, and control. This is a fun piece and should not be encumbered with stylistic dogma, so I'm not at all opposed to this alternative view of the score.

As for the sound, when you have three of anything, you inevitably end up with a first, second, and third preference, and from that point of view this is my third choice here as an audio demo. But if this were the only recording of the New York/Mehta forces by Max, I'd still be blown away and raving—it's good enough for that.

Sibelius

Jean Sibelius: Symphony No. 2 in D Major, Op. 43; Finlandia, Op. 26. New York Philharmonic, Zubin Mehta, conductor. ***Teldec*** *2292-46317-2 (DDD, recorded January 1990, released 1990).*

The symphonic fabric of Sibelius is mosaicked rather than woven; it works by juxtaposing contrasting material without much join. I think Mehta is very comfortable in this idiom, and interpretively this is probably the top-ranking CD in the series. I absolutely wallow in the socko finale of the symphony, corny and repetitious as it may be; I crank up the volume and conduct it. The same with "Finlandia," the perfect ten-minute audio demo. Go, Zubin, go!

This is a live to two-track recording, like the Hoist, and it proves how well that can work even with large forces. Stupendous brass, superbly detailed bass, great presence, yet with a panoramic touch of mellowness. Well done, Max.

And on Other Labels...

I have a lot of recent releases to cover here, so I'll try to be brief but enlightening.

Bach

"The Organ Works of J. S. Bach: Volume 1" (16 selections). Jean Guillou, at the Kleuker organ of the Eglise Notre-Dame des Neiges, Alpe d'Huez, France. ***Dorian*** *DOR-90111 (DDD, produced and recorded 1987 by Craig Dory, released 1990).*

This came out of the same November 1987 sessions as the marvelous "Organ Encores" I raved about in Issue No. 13. My favorite organist playing my favorite organ and recorded by one of my three or four favorite recordists—what else is there to say? That Bach was the greatest-ever composer for the organ? If you don't know that already, then you *really* must get this CD. As soon as the incomparable Jean Guillou plays the opening theme of the Prelude and Fugue in A Minor on track 1, it's obvious that something other than the usual earnestly plodding organ performance is about to take place. (Glenn Gould's opening bars of a Bach piece on the piano used to make the same impression.) A Guillou/Dorian version of the entire oeuvre of Bach for the organ is a prospect almost too good to be true.

J. *S. Bach: Mass in B Minor. Atlanta Symphony Orchestra & Chamber Chorus, Robert Shaw, conductor; Sylvia McNair, soprano; Delores Ziegler, soprano; Marietta Simpson, mezzo-soprano; John Aler, tenor; William Stone, baritone; Thomas Paul, bass.* ***Telarc*** *CD-80233I2CD (DDD, produced by Robert Woods, recorded 1990 by Jack Renner, released 1990).*

Robert Shaw can always be counted on for a good, solid performance and occasionally for an inspired one. To me this is in the good, solid category. The enlivening inflection that makes certain Bach performances special is in short supply here. Shaw opts for alternating concertists and ripienists in the choral passages, an "authentic" practice facilitated by his excellent soloists, but his beat is almost too reverential—Bach's got rhythm, man, even in his sacred music. Jack Renner's choral recordings are invariably gorgeous, and this is no exception.

Beethoven

Ludwig van Beethoven: The Sonatas for Piano, Vol. 4. Bruno-Leonardo Gelber, piano (Steinway). Sonata No. 21 in C Major, Op. 53 ("Waldstein"); No. 27 in E Minor, Op. 90; No. 32 in C Minor, Op. 111. ***Denon*** *CO-74653 (DDD, produced by Takashi Baba, recorded 1989 by Peter Willemoës, released 1990).*

I remember the Argentine pianist Bruno-Leonardo Gelber from his first American tour back in the 1960s, when he made a very favorable impression as a serious young artist. Here he plays excessively "big time," as if he needed to remind everybody that he is a world-class virtuoso. I prefer a more introspective, less explicitly assertive approach to Beethoven, whose assertiveness is built into the music and is evident without underscoring. Richard Goode, for example, who doesn't quite have the chops of a Gelber, comes closer to my Schnabel-influenced ideas of how this music goes, although Gelber is certainly far from negligible as a performer. The recording is very realistic in a cold, percussive way; it's a close-miked B&K job. Overall, a good but not great CD.

Berlioz

Hector Berlioz: Te Deum, Op. 22. Frankfurt Radio Symphony Orchestra & Choruses, Eliahu Inbal, conductor; Keith Lewis, tenor; Matthias Eisenberg, organ. ***Denon*** *CO-76142 (DDD, produced by Yoshiharu Kawaguchi and Richard Hauck, recorded 1988 by Detlev Kittler, released 1990).*

Less well known and celebrated than the Requiem, this later work of Berlioz has moments of comparable beauty, grandiosity, and originality. Check out the last two movements—a lovely prayer for tenor solo and women's chorus, followed by an overpowering, gloriously theatrical finale for multiple choruses and augmented orchestra, with lots of percussion. As always when Inbal is in charge, every small detail is audible and in balance with everything else. The recording (Alte Oper in Frankfurt, as usual) is quite wonderful, possibly the finest example I've heard so far of Denon's special technique using a B&K omni pair supplemented with digitally delayed B&K cardioids. Good show.

Brahms/Webern

Johannes Brahms: Symphony No. 2 in D Major, Op. 73. Anton Webern: Im Sommerwind (Idyll). Royal Concertgebouw Orchestra, Riccardo Chailly, conductor. ***London*** *430 324-2 (DDD, produced by Andrew Cornall, recorded 1989 by John Dunkerley, released 1990).*

The Brahms Second in the wrong hands can be quite boring, at least to this listener. It doesn't "play itself like, say, the Beethoven Seventh. Those long, leisurely melodic lines and the dark-hued orchestration require very clear-headed, meticulous, illuminative conducting. Chailly delivers the required goods. I find his performance thoroughly satisfying and wouldn't want to change a single bar of it. It's totally transparent, unaffected, and bracingly upbeat—a highly musical "reading" instead of an "interpretation." The Royal Concertgebouw plays magnificently. The bonus piece is a pleasant journeyman exercise in the late-romantic vein, composed by Webern when he was barely out of his teens and before he came under the influence of Schonberg. There's nary a hint in it of the dodecaphonist-to-be. The recording is one of the finest examples of the Decca/London multimiked approach, without a trace of the zingy quality I sometimes fault them for.

Johannes Brahms: Piano Concerto No. 1 in D Minor, Op. 15; Tragic Overture, Op. 81. Royal Philharmonic Orchestra, Andre Previn, conductor; Horacio Gutierrez, piano. **Telarc** *CD-80252 (DDD, produced by Robert Woods, recorded 1990 by Jack Renner, released 1991).*

Walthamstow Town Hall in London is one of the best possible venues for an orchestral recording. Here it adds a lovely warmth and resonant depth to Jack Renner's familiar Schoeps omni sound—but wait a minute, according to the technical notes he is now using Schoeps hypercardioids as well and, believe it or not, a good old-fashioned Neumann M-50. (Can't we take anything for granted anymore?) No matter, the sound is absolutely beautiful and the first thing that grabbed my attention. Even today, few CDs are this good. Then I noticed that the performance by Gutierrez and Previn is a very classy one, my delayed reaction being due to the limited affection I bear for the concerto, which is a bit too portentous for its contents—to my opinionated ear. The overture, on the other hand, is in the same league with Brahms's better symphonic first movements and fares equally well under Previn's baton. There's not a thing wrong with this CD except that it has too much first-rate competition in the catalog.

Chopin

Frederic Chopin: 24 Mazurkas. Charles Rosen, piano. **Globe** *GLO 5028 (DDD, produced and recorded 1989 by Klaas A. Posthuma, released 1990).*
Frederic Chopin: Sonata for Piano in B Minor, Op. 58; Sonata for Cello and Piano in G Minor, Op. 65. Charles Rosen, piano; David James, cello. **Globe** *GLO 5026 (DDD, produced and recorded 1989 by Klaas A. Posthuma, released 1990).*
Frederic Chopin: Polonaise-fantaisie in A-flat Major, Op. 61; Sonata No. 2 in B-flat Minor, Op. 35; Ballade No. 1, Op. 23; Ballade No. 3, Op. 47; Barcarolle in F-sharp Major, Op. 60. **Music & Arts** *CD-609 (DDD, produced and recorded 1989 by Judith Sherman, released 1990).*

Charles Rosen is not only a first-rate pianist and highly cultivated musician but also a music critic of considerably greater sophistication than I'll ever be, so I'm in a no-win situation here: I can't possibly tell him how *else* he should play Chopin. As a lifelong Chopin enthusiast I'll venture an opinion, however: Rosen's top priority appears to be structural clarity, whereas my top priority in Chopin is expression, or call it feeling. For expression/feeling Artur Rubinstein is my model in this music, and I find Rosen's impressively articulated performances less captivating than Rubinstein's. It's possible that, if the printed editions of the music were somehow lost, a scholarly researcher would find it easier to recreate them from Rosen's playing than almost anybody else's, but my understanding is that Chopin himself didn't play that way. Anyway, the recordings here are all very good, the Globe piano sound being a little rounder and richer, the Music & Arts more percussive and "trebly."

Diamond

David Diamond: Symphony No. 4; Concerto for Small Orchestra; Symphony No. 2. Seattle Symphony (in the Symphonies No. 2 and 4), New York Chamber Symphony (in the Concerto), Gerard Schwarz, conductor. **Delos** *DE 3093 (DDD, produced by Adam Stern, recorded 198911990 by John Eargle, released 1990).*

David Diamond, now in his mid-70s, is just beginning to be acclaimed as a great composer, partly because of this recording. Arnold Schonberg once called him "a new Bruckner" and tried to dissuade him from paying any attention to 12-tone technique. Not to worry, Arnie—Diamond has remained about as tonal as a 20th-century composer can be, and why not? A hundred years from now, who will care that his Second Symphony, written in 1942-43, sounds more like 1907? It happens to be a stunning, wonderful-sounding piece of large-scale orchestral music, and its date of birth is largely academic. The Bruckner comparison isn't so farfetched; Diamond's structures are also blocky and stop-and-go, but his sonorities and lyricism are irresistible. The recording is in the very best of the now familiar John Eargle panoramic-yet-detailed idiom—meaning somewhere in the neighborhood of state of the art—and the bass drum, especially, is awesome. You've got to hear this.

Dvorak

Antonin Dvorak: Slavonic Dances, Op. 46 and Op. 72. The Cleveland Orchestra, Christoph von Dohndnyi, conductor. **London** *430 171-2 (DDD, produced by Paul Myers, recorded 1989 by John Pellowe, released 1990).*

This is lovely, colorful music, replete with rhythmic and melodic delights, Op. 72 even more than Op. 46. The Cleveland Orchestra is of course a world-class outfit, and Dohnanyi is a superb conductor, perhaps not as relaxed and unbuttoned here as the music could well stand but still very effective. What I don't like about this CD is the recording. It's clean and dynamic but confused in spatial structure—too many microphones?—and quite aggressive in the upper midrange and highs. You can't listen to all 74 minutes of it without experiencing fatigue. Telarc, with Jack Renner, used to do an incomparably better job with the same orchestra in the same Masonic Auditorium.

Antonin Dvorak: Symphony No. 9 in E Minor, Op. 95 ("From the New World"); Carnival Overture, Op. 92. Los Angeles Philharmonic Orchestra, Andre Previn, conductor. **Telarc** *CD-80238 (DDD, produced by Robert Woods, recorded 1990 by Jack Renner, released 1990).*

Here the recording is Jack Renner's best (and I see for the first time, in the technical credits, a B&K 4011 cardioid rounding out his omni array), but the performance of the symphony is merely competent next to—among others—the superbly idiomatic one by the Czech Philharmonic Orchestra under Vaclav Neumann on Supraphon (almost as well recorded at the dawn of DDD, 8 ½ years earlier, maybe not quite as transparently). They just can't Czech it out in Southern California the way they can in Prague. The "Carnival Overture" is rousingly played on the Telarc disc, by the way.

Grofé/Copland

"Out West: Tone Poems of the American West." Ferde Grofé: Grand Canyon Suite. Aaron Copland: Billy the Kid (Suite from the Ballet); Rodeo (Four Dance Episodes). Seattle Symphony, Gerard Schwarz, conductor. ***Delos*** *DE 3104 (DDD, produced by Adam Stern, recorded 1990 by John Eargle, released 1991).*

The Grofé suite isn't much more than glorified movie music, but the Copland pieces are American classics—and if there's anything Gerard Schwarz knows how to conduct it's an American classic. These are beautifully lucid, carefully molded, idiomatic performances; to my ear they leave nothing to be desired. The recording reflects the most recent John Eargle orchestral microphoning techniques, than which there's nothing better; the gunfight sequence from "Billy the Kid" will be an audio demo piece for years to come. Bravo!

Handel

George Frideric Handel: 12 Concerti Grossi, Op. 6.1 Solisti Italiani. ***Denon*** *CO-76305I6I7 (DDD, produced by Takashi Baba, recorded 1989 by Peter Willemoës, released 1990).*

I Solisti Italiani are the successors to / *Virtuosi di Roma* and represent string playing in the finest Italian tradition. They play this magnificent music—one of the cornerstones of the baroque chamber repertory—with tremendous verve, considerable stylistic authority, and great beauty of tone. This is music making of a high order. The recording with B&K microphones is perfectly natural-sounding and presents a soundstage of just the right width and depth. Highest recommendation.

Janácek

Leos Janácek: Sonata—October 1, 1905 ("From the Street"); On an Overgrown Path, Books 1 and 2; A Recollection; In the Mist. Rudolf Firkusny, piano. ***RCA Victor Red Seal*** *60147-2-RC (DDD, produced by David Frost, recorded 1989 by Paul Goodman, released 1990).*

No one is better qualified than Rudolf Firkusny to play the deceptively simple but rhythmically and coloristically extremely subtle piano music of his boyhood teacher, Leos Janácek. Firkusny is able to combine keyboard control and emotional expression into a fluid playing style that gives endless satisfaction. The recording is on the conservative side—slightly soft-focus and not as clangorous as some—but still has more than sufficient presence and suits the gently impressionistic, introverted quality of the music very well.

Mahler/Zemlinsky

Gustav Mahler: Symphony No. 6 in A Minor. Alexander Zemlinsky: Sechs Gesdnge nach Maeterlinck, Op. 13. Royal Concertgebouw Orchestra, Riccardo Chailly, conductor; Jard van Nes, mezzo-soprano (in the Zemlinsky). ***London*** *Set 430 165-2 (DDD, produced by Andrew Cornall, recorded 1989 by John Dunkerley, released 1990).*

Mahler's Sixth is one of his works that I've always had trouble relating to, despite its high current standing among critics. There are some undoubtedly beautiful things in it, but all that gloom and doom, the ranting and raving, the interminable finale with its hammer blows (is it hammer time yet?) create a sense of surfeit in my musical viscera. That probably disqualifies me from distinguishing a good performance from a great one, so I can merely report that the Royal Concertgebouw plays superbly and that Chailly makes everything sound crystal clear. Two distinguished critics who have already reviewed this CD called the finale the strongest and the weakest part of the performance, respectively, so I'm in good company with my ambivalence. The recording is perhaps the best example of Decca/London multimiking I've encountered so far; it will please many with its brilliance and high resolution, but I still prefer Max Wilcox's warmer, weightier, and more simply microphoned Mahler sound, as reviewed above. As for the six orchestral Maeterlinck songs by Zemlinsky, Mahler's somewhat younger Viennese contemporary and Schonberg's teacher, you'll like them if you like early, tonal Schbnberg, as I do.

Mozart

Wolfgang Amadeus Mozart: Symphonies No. 25 in G Minor, K. 183; No. 32 in G Major, K. 318; No. 33 in B-flat Major, K. 319; No. 40 in G Minor, K. 550; No. 41 in C Major ("Jupiter"), K. 551. The Philharmonia Orchestra, Emmanuel Krivine, conductor. ***Denon*** *81757 6103 2 for Nos. 25 and 40;* ***Denon*** *81757 6579 2 for Nos. 32, 33, and 41 (DDD, produced by Yoshiharu Kawaguchi, recorded 1988/1989 by Hiroshi Goto, released 1990).*

Assuming the unlikelihood that you don't own the great G Minor and "Jupiter" symphonies in any form, should you then go out and get the Krivine versions? I wish the answer were a simple yes or no. The Philharmonia Orchestra is a distinguished group that plays with stringquartetlike refinement and flexibility, and Krivine's ideas about this music are interesting and plausible. But what about Toscanini, Walter, Beecham, Klemperer, Furtwangler, Reiner, Szell, etc.—all the great conductors whose interpretations are now available on CD transfers? Well, consider the sound. These B&K-miked recordings by Nippon Columbia are of remarkable beauty and transparency. You hear everything. So what will it be—100% of a very good conductor or 50 to 60% of a great one? Your choice alone.

Wolfgang Amadeus Mozart: Piano Concertos No. 21 in C Major, K. 467, and No. 27 in B-flat Major, K. 595. John O'Conor, piano; Scottish Chamber Orchestra, Sir Charles Mackerras, conductor. ***Telarc*** *CD-80219 (DDD, produced by James Mallinson, recorded 1989 by Jack Renner, released 1990).*

Two of the greatest masterpieces of the piano concerto literature, played by an outstanding artist and recorded on one of the truly audiophile-oriented labels—what a feast is promised here! Well, it turns out to be a little less scrumptious than that. O'Conor sounds like something of a cold fish in this music; I like him better in Beethoven. Mozart ought to effervesce more in the fast passages and sing more in the slow ones; O'Conor just sort of rococoes along in a musicianly way. Mackerras is a good musician, too, and the orchestra is fine, but the violins are recorded a little too close and tend to screech a bit in the high passages. Maybe it's the Glasgow City Hall acoustics, but then I see that Robert Woods is not the producer this time—could that be the reason?

Poulenc/Rachmaninoff

Francis Poulenc: Mass in G Major; Motets for Christmas and Lent; Four Short Prayers of Saint Francis. Robert Shaw Festival Singers (Emory Institute, Quercy, France), Robert Shaw, conductor; Donna Carter, soprano (in the Mass); Christopher Cock, tenor (in the Prayers). ***Telarc*** *CD-80236 (DDD, produced by Robert Woods, recorded 1989 by Jack Renner, released 1990).*
Sergei Rachmaninoff: Vespers (All-Night Vigil), Op. 37. Robert Shaw Festival Singers (Emory Institute, Quercy, France), Robert Shaw, conductor; Karl Dent, tenor. ***Telarc*** *CD-80172 (DDD, produced by Robert Woods, recorded 1989 by Jack Renner, released 1990).*

I'm lumping these two very different CDs together because they're so very similar in every respect except the music. Both were recorded at the same time, in the same place (Church of St. Pierre, Gramat, France), with the same performers singing a cappella, by the same recording team, with the same equipment (B&K 4006 omnis were used exclusively). Their sound is almost the same but not quite: the Poulenc has a very slightly more aggressive top end; the Rachmaninoff is sheer perfection (it received a Grammy for engineering). Interestingly, the Poulenc was recorded last; perhaps someone decided that the mikes could be moved in just a tad for even better definition—and he was wrong. The difference in music is very great, however; the Poulenc is pungent, varied, full of surprises; the Rachmaninoff is a steady, measured outpouring of superb Byzantine gloom. Each is masterful in its own way, and Shaw is masterful in both. He never sacrifices clarity and shape for beauty of sound, nor vice versa. To borrow from those Midas commercials, nobody beats Shaw at this sort of thing. Nobody.

Rimsky-Korsakov

Nikolai Rimsky-Korsakov: Scheherazade, Suite symphonique, Op. 35; Capriccio espagnol, Op. 34. London Symphony Orchestra, Sir Charles Mackerras, conductor. ***Telarc*** *CD-80208 (DDD, produced by James Mallinson, recorded 1990 by Jack Renner, released 1990).*

Six months after the okay-but-far-from-great Mozart concerto job reviewed above, the exact same team in England, using very similar equipment, made this stupendous-sounding recording. What made the difference? Rimsky? No way. It had to be the hall, Walthamstow Town Hall in London, one of the prime recording venues in the Western world. I'm sure that the fortissimo trombone passages of "Scheherazade" are already in use as an audio test in various circles—they're awesome. This is pure Schoeps sound, and there's not a trace of hardness on top of that you-are-there brass. The London Symphony Orchestra plays beautifully, and Mackerras shapes and illuminates these war-horses with tender loving care, as if they were great music. I don't see how any red-blooded audiophile can pass this one up.

Schubert

Franz Schubert: "The Complete Works for Violin and Piano." Sonatina in D Major, D. 384 (Op. 137, No. 1); Sonatina in A Minor, D. 385 (Op. 137, No. 2); Sonatina in G Minor, D. 408 (Op. 137, No. 3); Sonata ("Duo") in A Major, D. 574 (Op. 162); Rondo ("Rondo brillant") in B Minor, D. 895 (Op. 70); Fantasy in C Major, D. 934 (Op. 159). Jaime Laredo, violin; Stephanie Brown, piano. **Dorian** *DOR-90137 I, II (DDD, produced by André Gauthier, recorded 1989 by Craig Dory, released 1990).*

Schubert never wrote a towering masterpiece for the combination of violin and piano, but all of these works are suffused with his unique melos, and the world of music would surely be shortchanged without them. One must add that the 26-minute Fantasy in C Major is definitely a major work, with passages of great beauty, but not quite the peer of the symphony and quintet in the same key of the same period. As for the performers, I'm more comfortable with Stephanie Brown, a very solid musician, than with Jaime Laredo, whose tone leaves something to be desired from time to time. Somehow they don't come off as the dual expression of a single musical intelligence, although much of the playing is quite excellent. The recording is exactly what I expect from Craig Dory in the Troy Savings Bank Music Hall—vividly clear and utterly natural.

Franz Schubert: Piano Quintet in A Major, D. 667 ("Trout"); Quartet No. 13 in A Minor, D. 804. Cleveland Quartet (William Preucil , violin; Peter Salaff, violin; James Dunham, viola; Paul Katz, cello); John O'Conor, piano; James VanDemark, bass. ***Telarc*** *CD-80225 (DDD, produced by James Mallinson, recorded 1990 by Jack Renner, released 1990).*

If I were asked to choose a CD for the purpose of introducing a rank novice to the delights of chamber music, I could do a lot worse than to suggest this one. The music is of universal appeal, not the least bit weighty but enchantingly beautiful even after the 100th hearing; the performances are simply lovely (this is not the "cold" O'Conor I was talking about), and the recording is smooth as silk. Truly a model CD, except that chamber-music lovers will already have their nonnegotiable "best" version of each work on the shelf. And by the way—would you believe it?—Jack Renner used an old Schoeps *tube* model to record this with. That's one for the tubes-sound-better contingent—of course, there are other reasons for the excellent sound of this production, but why not let 'em have their little fun?

Schumann

Robert Schumann: Carnaval, Op. 9; Papillons, Op. 2; Toccata, Op. 7. Cecile Licad, piano. ***Sony Classical*** *SK 45742 (DDD, produced by Gary Schultz, recorded 1989 by Bud Graham, released 1990).*

Schumann, Chopin, and Liszt were almost exactly the same age, and between them ended up with a near monopoly of the early romantic piano repertory. Schumann's position within that repertory is defined mainly by his early compositions; these pieces were all composed in his early twenties and are classics—late Schumann is not as reliable. Cecile Licad, an extremely talented young Filipino artist, plays this music just a notch below the exalted level of a Rachmaninoff or a Rubinstein, in other words as beautifully as you're likely to hear anywhere today. I'm very impressed by her subtle musicianship and virtuosity; she is definitely going places. The recording is of the school that wants to put the piano in your listening room—very little hall sound but great realism in terms of tonality and dynamics. In my large room that works just fine.

Shostakovich

Dmitri Shostakovich: Symphony No. 10 in E Minor, Op. 93. Atlanta Symphony Orchestra, Yoel Levi, conductor. ***Telarc*** *CD-80241 (DDD, produced by Robert Woods, recorded 1989 by Michael Bishop and Robert Woods, released 1990).*
Dmitri Shostakovich: Festive Overture, Op. 96; Symphony No. 10 in E Minor, Op. 93. Helsinki Philharmonic, James DePriest, conductor. ***Delos*** *DE 3089 (DDD, produced by Adam Stern, recorded 1990 by John Eargle, released 1990).*

There are those who consider the Tenth to be Shostakovich's "greatest" symphony; I'm not even sure if he was a "great" composer (in the sense of a Stravinsky or a Bartok), but I'll concede that it's a big, colorful, expertly wrought, impressive musical structure, highly listenable but not thrilling to this listener. These two recordings could be considered to be more or less on a par performancewise if it weren't for Levi's inexplicably slow tempo in the first movement, tipping the scales decisively in DePriest's favor. DePriest and the surprisingly excellent Helsinki orchestra are in general a little more psyched and focused in this work than Levi/Atlanta, although I could be quite happy with the latter in the absence of an alternative. The Delos disc has the further advantage of having room for the catchy Festive Overture as a bonus—very well played, too. Sonically both recordings are outstanding; Telarc without Jack Renner is still recognizably Telarc, and John Eargle does wonders with the smallish Helsinki hall; here, too, I tend to lean slightly toward the Delos version.

Sibelius

Jean Sibelius: Symphony No. 1 in E Minor, Op. 39; Symphony No. 5 in E-flat Major, Op. 82. Atlanta Symphony Orchestra, Yoel Levi, conductor. ***Telarc*** *CD-80246 (DDD, produced by Robert Woods, recorded 1989/1990 by Michael Bishop, released 1990).*

Here I have no reservations about Levi's conducting; indeed, I have yet to hear a clearly better performance of the Sibelius First by anyone, and the Fifth is almost as well played. Levi has mastered the quirks of the Sibelius idiom, the orchestra executes his wishes to a T, and the results are highly compelling. Add to that a nothing-but-Sennheiser omni recording—remember, I changed my opinion about that mike—and you have a valuable addition to any basic CD library, one that could even serve as a novice's introduction to Sibelius, with audio as the hook.

Smetana

Bedrich Smetana: The Complete Czech Dances (Book One and Book Two). Antonin Kubalek, piano. ***Dorian*** *DOR-90122 (DDD, produced by Douglas Brown, recorded 1988 by Craig Dory, released 1990).*

This is the musical equivalent of Czech dumplings—a little on the unsubtle side, very ethnic, and quite delicious. There's plenty of showy display and some very catchy tunes. Great stuff, all of it. Kubalek is perfectly cast here; I can't imagine anyone playing these pieces more idiomatically—the man obviously loves his dumplings and digs into them with gusto. The recording is a special treat, one of the most beautiful examples of piano sound in my entire CD collection. If this is the sound Craig Dory could get in the Troy Savings Bank Music Hall *2½* years ago, I wonder why he then started to experiment with a more reverberant characteristic.

Sousa

John Philip Sousa: The Original All-American Sousa! Keith Brion and his New Sousa Band (13 marches); John Philip Sousa with his band (historical, 7 marches). ***Delos*** *DE 3102 (DDD, produced by Adam Stern, recorded 1990 by John Eargle, released 1990).*

If you think Sousa's marches are supposed to sound the way your high-school or college band used to play them, this CD will be an ear-opener. Keith Brion's band is dedicated to the authentic Sousa performance style, and of course Sousa's own recordings (1917-29) document what that style should be—definitely not oompah or sizz-boom-bah. The juxtaposition, on the same disc, of all the available Sousa historicals with the superior John Eargle recordings of the Brion performances (RCA Studio A in New York City—neither the best nor the worst of venues) makes for a unique document and confirms what has always been my impression: the best symphonies may not be American but the best marches are.

Tchaikovsky/Rachmaninoff

Peter Ilyich Tchaikovsky: Piano Concerti No. 1 in B-flat Minor, Op. 23, and No. 3 in E-flat Major, Op. 75. Vladimir Feltsman, piano; National Symphony Orchestra, Mstislav Rostropovich, conductor. ***Sony Classical*** *SK 45756 (DDD, produced by Steven Epstein, recorded 1989 by Bud Graham, released 1990).*
Peter Ilyich Tchaikovsky: Piano Concerto No. 1 in B-flat Minor, Op. 23. Sergei Rachmaninoff: Rhapsody on a Theme of Paganini, Op. 43. Horacio Gutierrez, piano; Baltimore Symphony Orchestra, David Zinman, conductor. ***Telarc*** *CD-80193 (DDD, produced by Robert Woods, recorded 1990 by Jack Renner, released 1990).*

The world doesn't need a new recording of the Tchaikovsky piano concerto—the First, that is—unless it's quite special. The competent Gutierrez/Zinman version doesn't meet that criterion; the Feltsman/Rostropovich does. The latter has oodles of warm, expansive Russian soul, which is one very good approach to this war-horse. When that third-movement peroration comes, it has to bring tears to your eyes—or forget it. The Russians deliver the tears, and they also give you an equally effective performance of the much less frequently heard, one-movement Third concerto. The Rachmaninoff bonus on the Telarc disc again suffers from far too much world-class competition without being in any way negligible by itself. The audio quality in both recordings is just a tad below state-of-the art, i.e., more than good enough to be a nonissue in the comparison. The Telarc is perhaps a shade more refined in texture—it really doesn't change matters. The Sony release is a good introduction, by the way, to the highly publicized art of Vladimir Feltsman.

...and jazz:

Drummond/Jones/Higgins

"The Essence." Ray Drummond, acoustic bass; Hank Jones, piano; Billy Higgins, drums, ***dmp*** *CD-480 (DD, produced by Ray Drummond and Tom Jung, recorded 1990 by Tom Jung, released 1991).*

I'm no great fan of the kind of "contemporary," electrified, sound-effects-oriented jazz Tom Jung seems to favor for his dmp label, but—wow!—is this untypical! Fantastic! No, it isn't 1990s jazz, thank God; my late-'50s/early-'60s ear has no trouble relating to it. But why date it? It's superb, pure-acoustic jazz of the finest, most imaginative, most sophisticated sort. Ray Drummond is awesome; in fact, his earlier collaboration with the pianist Bill Mays *(One to One,* **dmp** CD-473), which I also enjoyed as an untypical Tom Jung release, didn't quite prepare me for this experience. Drummond's virtuosity on the acoustic bass provides textbook examples of the jazz possibilities of that instrument, so often submerged in the improvisational fabric. Hank Jones and Billy Higgins are also terrific musicians, and the whole session exudes effortless, understated mastery, the very definition of cool. The songs are mostly classics (Duke Ellington, Gershwin, Johnny Mercer, etc.), but the title tune by Ray Drummond is based on—believe it or not—a Bartok theme. The all-digital (including mixing console!) live-to-two-track recording by Tom Jung is of course sheer perfection, as usual. You can hear every wire in Billy Higgins's brushes. My kind of jazz CD.

Oscar Peterson Trio

"The Legendary Oscar Peterson Trio Live at the Blue Note." Oscar Peterson, piano; Herb Ellis, guitar; Ray Brown, bass; Bobby Durham, drums. ***Telarc*** *CD-83304 (DDD, recorded 1990 by Jack Renner, released 1990).*

This won a Grammy in February 1991, and I could have predicted it. Oscar Peterson at 65 shows no signs of slowing down; he is a transcendental swinger and bopper who holds your attention with every note. I think he is best in his own compositions here ("Peace for South Africa," "Sushi," "Blues for Big Scotia"), but the oldies on the disc are also great. Every now and again he lets loose with a burst of keyboard virtuosity that bowls you over. The live nightclub recording presents some problems; I don't mind the audience noises and applause, but the hollow, echoey acoustic bothers me here and there, and Herb Ellis's guitar riffs are lost in the backround, although he becomes very audible when he solos. Even so, this is probably the highest-fidelity Oscar Peterson recording ever, just by virtue of Jack Renner's hardware and recording technique. 0

By concentrating on precision, one arrives at technique; but by concentrating on technique one does not arrive at precision.

—BRUNO WALTER

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The Audio Critic

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Dr. Rich does his professorial number on an assortment of audiophile preamplifiers, with evaluations of circuit design and parts quality per dollar, plus in-use tests, etc.

We publish the promised comprehensive article on the various methods to obtain deep bass from small speaker boxes (dropped from this issue for lack of space).

We continue to review all the speakers we can get our hands on, including the unique new Win SM-10.

Wire/cable facts and fictions are further examined and elucidated, this time with the emphasis on interconnects.

More reviews of analog and digital electronics, including a sprinkling of high-end video, plus our columns.