Volume 1, Number 4 July/August/September 1977

# The Audio Critic

**In this issue:**

The speaker survey at last, with 15 systems from $224 to $5200 a pair compared in this first go-around. (The $5200 one sounded best, alas.)

Our power amplifier survey continues in a rather positive vein, especially since this second batch includes our new reference standard.

We launch what may be our most important and, to some, most disturbing inquiry so far: an updated investigation of the cartridge/arm/turntable relationship. Including 10 tone-arm and turntable reviews for openers.

Plus, of course, our regular features.

The Audio Critic

Vol. 1, No. 4 July/August/September 1977

Editor and Publisher Peter Aczel
Associate Editor Max Wilcox
Graphic Designer Dick Calderhead
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Assistant to the Publisher Elizabeth Tinsley

Consulting engineers and other technical advisers are engaged on a project basis, some contributing under their by-lines, others working anonymously.

**The Audio Critic** is an advisory service and technical review for consumers of high-priced audio equipment. It is published six times a year by The Audio Critic, Inc., and is available by subscription only. To maintain total dedication to the consumer's point of view, **The Audio Critic** carries no advertising by equipment manufacturers, distributors, reps, dealers or other commercial interests. Any conclusion, rating, recommendation, criticism or caveat published 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, Box 392, Bronxville, New York 10708.



**For subscription information and rates, see inside back cover.**

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# Publisher's Note

*In our Number 2 issue (March/April 1977), we made the following ill-considered editorial statement: ". . . Then we'll get out three more issues in the second half of the year. If it's October and you haven't received Number 4 (July/August) yet, then you'll have cause to worry—we won't be able to catch up. But it isn't going to happen."*

*Well, October has come and gone, and you haven't received . . . but wait a minute, you're holding Number 4 in your hand, so it couldn't be as bad as all that. Of course, it's perfectly true that we won't be able to catch up—not in 1977. When we wrote those words in April, we sincerely believed that we could publish three times as many issues as any other audio review that carries no advertising (the shortest interval we had ever observed between two successive issues of these having been six months). Well, in the sober gray light of October, it appears that we're able to publish only twice as many issues as anyone else in our business, our average interval thus far being three months. That doesn't mean, however, that we have no potential for improvement. Here's our plan:*

*To reflect reality, we're dating this issue to cover three months: July/August/September 1977. (We'll probably do the same thing in 1978, since the summer seems to be inevitably our period of lowest output.) Our Number 5 issue will be dated October/November 1977, and we're reasonably confident that we can mail it out in December. That leaves Number 6, which will be dated Year-End 1977 and (you guessed it) won't be mailed out before we're well into 1978. There's just no other way. Number 6 will be worth waiting for, however; we plan to make it a very special reference issue that audio enthusiasts will be brandishing around throughout the coming year, making life difficult for their friends and local dealers. After that, we'll see. Maybe we'lll have our act together to the point where we can have our twelfth issue out before the end of 1978. But this time we won't promise . . .*

*So the question, from our subscribers' point of view, is this: What's the difference between spending $28 for six issues over a period of twelve months and spending $28 for the same six issues over a period of, say, fourteen or fifteen months. If you feel that the first is a good deal and the second is a bad deal, then you have a legitimate quarrel with us. We think it's exactly the same deal, since we're selling you six issues either way, not a time-limited contract. We were, however, unrealistic when we committed ourselves to a time limit, and for that we owe you an apology.*

*One more thing. So far, we've religiously answered every where's-my-last-issue-I-didn't-get-it-yet letter. As of the day this issue is being mailed, we're ending that time-wasting and costly practice. Since 99.8% of our mailings reach their destination (that's an exact figure), please consider no reply to that kind of inquiry to mean: "We haven't mailed it yet."*

*Thank you.*

# Box 392

**Letters to the Editor**

*In addition to editorial correspondence of general interest, we're including two special categories of letters this time. One has to do with clearing the air after a particularly foul-smelling anonymous communication circulated about **The Audio Critic**; the other covers exhaustively the "Mark Davis syndrome" originally brought up in our first issue as a kind of aside in Part I of the preamp survey. The Mark Davis correspondence is reprinted here in 8-point type for those who are sufficiently motivated to plow through it; we didn't feel we'd be justified giving it more space. The letters we publish in this column 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, Box 392, Bronxville, New York 10708.*

*In the last week of June and throughout most of July, the anonymous missive reprinted below was being sent out to literally hundreds of high-end audio dealers, manufacturers, reviewers, editors and other high-end audio practitioners from coast to coast. The anonymous commentator had invested 24 cents first-class postage per mailing, as each fat envelope also contained Xeroxed copies of our allegedly misleading ads and of trade-press clippings documenting that two of our part-time staff consultants made their daily living in the audio business, one in retailing, one in manufacturing. All of these envelopes were without a return address and postmarked either in Hartford, Connecticut, or Manchester, Connecticut (a suburb of Hartford).*

The Audio Critic—A Ripoff?

Another "little magazine" has appeared.

This one bears watching.

Initially, our hopes were high. The price ($28.00 for a year's subscription) was high but then so was the promise. By combining objective measuring techniques with long term listening sessions, The Audio Critic promised information, not just opinion.

It now appears that the price was too high as The Audio Critic is, in our opinion, incapable of fulfilling the promise.

Contrary to claim, The Audio Critic has a number of substantial ties with the audio industry, both on the retail and manufacturers' level. These ties make objectivity impossible. In fact, these ties just might make The Audio Critic a "house paper."

The attached information begins to point out the connections.

We feel there may be more tidbits yet to be unearthed.

But even if these suspected alliances prove to be untraceable, The Audio Critic has misrepresented itself to the audiophile-consumer. Where is the line between misrepresentation and fraud?

Just as important—can anything written by The Audio Critic be believed?

We think not.

The Audio Critic is, to us, severely compromised. A publication with no credibility serves no other purpose than to make someone a lot of money.

Peter Aczel, the Editor and Publisher, has a lot of answering to do.

*In the wake of the anonymous mailings, the following exchange of letters took place:*

Dear Mr. Aczel:

No doubt you have seen the anonomous *(sic)* letter that is being circulated, accusing your publication of having "substantial ties with the audio industry," and supplying what appears to be some documentation of that allegation.

We were of course sent the sheaf of papers in question, and while we do not intend to print the letter—it goes rather overboard, in my opinion—we cannot let the matter pass without at least some comment to the effect that the letter was circulated and commercial affiliations of two of your key staffers documented from other, independent sources.

May I please have a statement of your response to this, for publication?

Sincerely,
J. Gordon Holt
Editor & Publisher
The Stereophile

*Dear Mr. Holt:*

*Two things surprise me about your recent letter.*

*One is that you assume you're* entitled *to a reply, even though you're neither a subscriber of this publication nor a manufacturer whose equipment has been reviewed in it. I'll let that pass.*

*The other is that you, as a reputable person, should be impelled to initiate this dialogue by something as sleazy as an anonymous letter, which should have been disposed of in the same manner as bird droppings. Since the same information is available to you from other sources, as you say, I wish you had referred only to the latter. It would have been cleaner that way.*

*It seems, however, that in addition to his soiled character, the anonymous letter writer also has a reading problem. As the enclosed excerpt from the second issue of The Audio Critic shows, I had already discussed quite freely in print what he started to whisper behind my back months later. (Moral turpitude and lack of mental alertness frequently go together.)*

*To round out this editorial on the subject (which I must ask you to reproduce in its entirety along with this letter, otherwise I can't give you my permission to print either one), I would also like to make the following points:*

*My wife and I are sole stockholders of The Audio Critic, Inc., which banks the subscription fees. Neither one of us has any affiliation with audio manufacturers or retailers, nor does anyone else on our payroll. The two* consultants *(not key staffers, as you say) triumphantly tracked down by the anonymous creep are both old friends of mine who had lent their names to my infant enterprise without any remuneration. Now that The Audio Critic is a going concern, I am entering, and have entered, into many more consulting relationships with technologists of all sorts employed by the audio industry—so many, in fact, that it's no longer practical to list them on the masthead, especially since some of them don't wish to be so listed.*

*I spend hours on the telephone or in person with these engineers and designers; I pick their brains; they pick mine; they influence my technical thinking; I influence theirs; they come to my laboratory; I go to theirs; we discuss test methods; we have drinks or even dinner together; they show me their latest research and papers —and they haven't got a nickel's worth of influence on my income or my editorial policy. The list of their names would excite the anonymous letter writer into an orgasm of I-told-you-so.*

*I, on the other hand, feel that such alliances are better for my education and therefore better for my subscribers than if I consorted exclusively with untutored audio freaks—which is obviously what the anonymous letter writer thinks an honest noncommercial audio journalist should do.*

*Does this answer your question?*

*Sincerely,*
*Peter Aczel*
*Editor and Publisher*

*(The editorial excerpt from our second issue, as enclosed with the above reply, is reprinted here.)*

*We have come under criticism by about 0.1% of our readership (three persons, to be exact) for the professional involvement of some of our staff consultants in the audio industry. Doesn't that constitute a conflict of interest, we were gleefully challenged, in view of our simon-pure posture of independence?*

*Ah, that's a good one. The basic reason for the amateurishness of the "underground" audiophile reviews is that they are staffed by amateurs. It would be very nice if one could come to valid conclusions about, say, the transient response of an amplifier by consulting music-loving dentists, accountants and shoemakers. Unfortunately, such independent experts seldom know what they're talking about. That's why we have a professional record producer, a physicist/audio engineer, an audio-electronics technician and other qualified professionals on our staff. Sure, some of these people derive part of or all of their income from the audio business, but not one of them is a chief executive officer or majority stockholder, so that the worst that can happen is that the views they privately communicate to us deviate from the self-interest of their bosses. Tough. It just so happens that* ***The Audio Critic*** *has already dealt rather severely with products made or sold by said bosses.*

*The important thing is that the* management *of* ***The Audio Critic*** *is completely divorced from commercial audio. The Editor/Publisher deliberately severed all connections with the industry before coming out with the first issue. Our subscribers are our sole business interest.*

*As a matter of fact, if you hear any malicious gossip about* ***The Audio Critic****'s conflicts of interest, or especially about our taking bribes for favorable reviews (one of the ever-recurring fabrications about nearly all audio reviewers, perpetuated by a few pathetic little would-be authorities), we suggest you let us know, provided you're willing to identify the source. There's nothing a Hungarian loves better than a good lawsuit.*

Dear Mr. Aczel:

We've had an anonymous packet regarding your publication's claim to impartiality. I imagine most publications in our field have. Would you care to comment on the present relationships of your staff to commercial interests?

Sincerely,
Edward T. Dell, Jr.
Editor/Publisher
The Audio Amateur

*Dear Mr. Dell:*

*In reply to your recent letter, I am enclosing a copy of a similar letter from J. Gordon Holt and of my response to it.*

*Since my answer to your letter, had it preceded Mr. Holt's, would have been virtually identical, there is really nothing further to say, except that this makes you the second reputable audio journalist without a clearly expressed distaste for anonymous letters.*

*Sincerely,*
*Peter Aczel*
*Editor and Publisher*
*The Audio Critic*

Dear Mr. Aczel:

We have received an anonymous note regarding your magazine, entitled "The Audio Critic—A Ripoff?" It's most probable that you have already seen a copy of this "material."

It appears that considerable time, trouble and expense went into the preparation and distribution of this note, but we are unable to determine the intent of the sender; altruistic self-righteousness seems out of the question due to the effort involved.

Some of us take issue with some of your published opinions, Mr. Aczel, but we all agree that your conclusions are fairly presented and without discernable bias. We question the composition of your reference system, having learned that there are amplifiers far more definitive than the Quatre, but we attribute that to subjective impressions rather than influence of industry.

You have our absolute support. We will continue recommending your fine publication to audiophiles within our area of influence. We are convinced that the more any client knows about audio, the better he'll like our way of doing business.

Sincerely,
H. L. Eisenson
Audio Dimensions
San Diego, CA

* * *

*And now for saner and more savory matters.*

The Audio Critic:

. . . I applaud your attempts to correlate lab data with audible effects; this field seems to be limited to yourselves and Richard Heyser in terms of published material, especially since Bascom King no longer reviews equipment publicly.

My major complaint with your magazine concerns a fault common to both you and your competitors, and is a simple one. Simply, there is no state-of-the-art in audio today (with the possible exception of tuners); to attempt to recommend one piece of equipment as the best of its art is foolhardy. To truly be the "state-of-the-art" a component must outperform its competition in every parameter considered. That is, when two pieces of equipment are compared, there should be no doubt in the minds of the listeners as to which is superior. If the sound of both components is felt to have different positive qualities, then clearly neither represents the SOTA, since existing technology could better either component. In fact, you allude to this characteristic several times, yet in the final analysis appear to ignore it. You could put yourselves yet another step above your competitors if you did not attempt to follow in their footsteps by finding a "best"; there is certainly more than one legitimate approach to the SOTA, and putting one above the rest smacks of the commercialism you are trying to avoid, as well as being unrealistic. . . .

Sincerely,
Robert Bertrando
Tucson, AZ.

*We agree with you wholeheartedly: a mixed bag of superiorities and inferiorities doesn't constitute SOTA, even if on balance the equipment is judged to be preferable to all others tested. We have tried to qualify our conclusions and recommendations to make that clear; unfortunately the word "best" tends to stick out and obscure qualifiers such as "so far" or "depending." We'll try harder in the future.*

*—Ed.*

The Audio Critic:

"Accuracy" is an impeccable word, and your bit on page 7 of the March/April issue, "In other words, accuracy isn't a matter of taste," is very neat and convincing. Up to a point.

As you say, "either they did or they didn't" match the color. Agreed also that it's difficult in audio, but your point remains valid, and taste has nothing to do with it. You look at your car and the repainted door and say, "They don't match." I don't doubt that you see a mismatch.

I come along. I look from car to door, and I say, "They did a fine job, didn't they!" *I* see a perfect match. "Accurate," I say. "Not accurate," you say. And we are both right.

We both see what we say we see. But we happen to have different eyeballs. I have a friend who sees almost no red, but sees beyond violet into ultraviolet, to make up for it; most variations are not so spectacular. Which is to say that your color-matching analogy is valid, your "accuracy" motto is valid—until another person comes along. Eyeballs are not standardized.

Ears are not standardized either.

So that impeccable word "accuracy" is not the solid support you may think it is. I hope you use it with care.

Sincerely,
Burnett Cross
Hartsdale, NY

*You have a point there but, in our opinion, you're bearing down a bit too hard on it. There exists a spread in physical perception from human to human, but between reasonably young and healthy specimens the spread is quite narrow, so that a norm for our species is a viable concept. If color matching, for example, were as subjective, indeed anarchic, as you suggest, wallpaper companies couldn't stay in business—nor could Eastman Kodak. In audio, we'll settle for a match to the original as good as the average car owner would let the body shop get away with.*

*—Ed.*

The Audio Critic:

. . . Would you please discuss the conflict (to my mind) between the ideal "straight wire with gain" concept, represented by the state of the art preamps and amps with a minimum of controls and frills, and the conflicting use of expanders, companders, equalizers, and noise reduction devices a la Phase Linear 1000 or Burwen, etc.

In other words, is it insane to own a Rappaport preamp, Quatre amp, DQ-10's, etc. *and* a ten-band/channel equalizer—dbx expander—SAE "click and pop machine"—Phase Linear 1000—or any one of a dozen accessories?

Do all the many add-on devices only serve to destroy the "art" in "state of the art" represented by an exceptional amp or preamp?

Thank you,
David Gibbs
Rancho Palos Verdes, CA

*The signal-processing accessories you mention are a necessary evil—and not quite as often necessary as a lot of audio people think. Such devices are designed to correct what somebody else has already messed up, either as a result of inadequate technology or through sheer neglect. With today's best program sources, played in a room that isn't an absolute pig acoustically, you're much better off with the shortest, simplest possible signal path from input to output. In fact, with all those "black boxes" in the signal path, you're quite right to suspect that the difference between, say, a Rappaport preamp and a Marantz would be obscured. We never touch the stuff ourselves, unless it's on loan for evaluation.*

*—Ed.*

The Audio Critic:

. . . There is one aspect of the preamp "controversy" which confuses me and, perhaps, other non-technical types. It is the question of stereo depth. If distortion gives the illusion of depth and true accuracy demands a sacrifice of the illusion, I would opt for accuracy. Isn't this question answered by using a record or tape where the reviewer would be aware of the precise location of the instruments during the live performance? A string quartet, even if the performers were local amateurs, might be ideal . . .

Yours truly,
Edward J. Doyle
East Northport, NY

*There's a subtle but audible difference between the haphazard artificial depth resulting from electronic anomalies and the focused natural depth rendered by accurate reproduction. There's no sacrifice involved in the latter; it sounds just right. We agree with you that complete familiarity with the depth and width of the sound source at the actual recording site is a great help in judging this phenomenon. Of course, the microphoning must be simple and straightforward; gimmicky multi-miking can throw you off completely.*

*—Ed.*

The Audio Critic:

Having purchased *two* Rappaport preamps on the basis of Vol. 1, No. 1, I must say your writers should have elaborated on the PRE-1's highs. They are objectively beautiful, but subjectively emotionally empty. Most people agree the program-listener schism is anxiety provoking. Has anyone built a non-fatiguing SP-4 with sock?

James A. Debros
St. Cloud, MN

*No, no, no! What you need is a nonobjectively transcendental preamp that is in perfect harmony with the vibrations of the universe, not to mention our precious bodily fluids.*

*—Ed.*

The Audio Critic:

I have just read the first two issues of The Audio Critic and found them to be very interesting. . . . As an electrical engineer I am happy to see perceptive criticism coming from persons who seem at the same time to believe that there are laws of physics and electricity which must hold no matter what our wishes to the contrary. I look forward to future issues and your continued success.

There are two comments that I would like to make regarding the following quotes from The Audio Critic, Vol. 1, No. 2. On page 41 you state:

"The net result is linear—without feedback and hence without TIM." There is at the present time some confusion about the precise term that should be used for the distortion now commonly called "Transient Intermodulation Distortion." Some persons call it "TIM," others call it Slew Induced Distortion or Transient Overload Distortion or Frequency Overload or any of several other terms. Whatever term is used, it is clear that this form of distortion can occur within a feedback loop or outside of any feedback loop.

Thus, it is not really correct to say that an amplifier, whether it consists of one or several stages, will certainly be free of transient distortion just because it has no feedback. A single stage amplifier with no feedback will fail the Otala test if its slewing capacity is exceeded. This is a small point but if we are ever to understand precisely what is going on in amplifiers we must think about them very carefully.

What is true about feedback amplifiers is that, if there is any tendency for internal slew or amplitude limiting to take place, the problem is greatly exacerbated by the feedback. However the problem *is not caused* by the feedback but *is caused* by the limiting tendency in the first place.

It is not correct to say "no feedback—thus no TIM." It is correct to say "little feedback—thus less problem with internal TIM."

My second comment relates to the following parenthetical expression: "(They'll never teach it this way at EE school, but then the tuition there is more than $28 a year.)"

I realize that this is just a nasty little comment set down to thrill your readers. (Critics are supposed to be mean to the core, I hear.) My issue with the comment is that it is not fair or accurate. In the attached reference list from an article by Jung, most of the references are from universities (Daugherty, Greiner, Leach) or certainly from highly trained and qualified electrical engineers.

We in the university are trying very hard to understand the very same problems that plague us all. I have for 15 years taught about overload problems in feedback amplifiers to my students. I have also written about these problems in the technical literature. The public and amplifier designers have not been listening very well.

Certainly the university is not the source of all knowledge, but we have tried. Much credit for improved amplifiers must go to the individual inventor of course. Unfortunately they have often been more reluctant to share their accomplishments and insights with the public.

What I am really suggesting to you is that we all need each other in order to both solve and understand the complexities of amplifiers of any kind, whether they are power amplifiers or pre-preamplifiers. In order to join together in these efforts we must make an attempt to communicate better. This will happen only if we try for harmony instead of acrimony.

Please consider these comments to be harmonious.

Let me close this time with a bit of acrimony of my own directed to the inventor-producer of some of the flood of equipment that exists.

It has been my observation that many writers, those that write for pulp magazines or the most esoteric journals, all have a primary purpose to spread their insight and understanding of problems to others. They are in fact teachers in the most generic terms. The editors of The Audio Critic are clearly doing the same. This is the way to understanding since the process gets the best minds focused on the problems.

On the other hand, the individual inventor-producer all too often has just the opposite goal in mind. He hides his techniques, he hides his theory, or worse, spouts garbage and nonsense; he misleads, he buries his circuits in epoxy, he gives erroneous specifications, he never writes about his circuits or theory of operation, he never supplies operational manuals that are correct, he never-never supplies repair manuals, and so forth.

This is not true of many manufacturers, of course, but all too often the most mystical manufacturers are the ones that make the most outrageous claims. It is very hard to tell the fakes from the frauds.

I would urge you to ferret out the fakes. Make them publish their theory, if any. Make them tell the truth about their specifications. Make them show the magic, if any, of their circuits. Have them explain how they defy the laws of physics.

I have hopes that you will accomplish more of this than the mystics among the golden-ear crowd have to date.

Very best regards,
R.A. Greiner
Professor of Electrical Engineering
University of Wisconsin

*We stand corrected on our TIM/feedback simplism and fervently endorse the admonishment of undisciplined inventor-producers and muddleheaded audio mystics. What distresses us, however, is that Professor Greiner, who is one of our heroes (he was coauthor of a superb 1966 paper that anticipated the entire Otala-Curl-Leach-Jung scene), should so direly misinterpret our little quip about EE school and the $28. We were actually apologizing, in our own left-handed way, about the brevity and patness of our explanation of the analog multiplier ("gain cell") principle. You academicians must really feel beleaguered if you take a mildly embarrassed journalist's foot shuffling to be a kick in the pants!*

*—Ed.*

The Audio Critic:

Re your CES report on the Verion NF-1 Filter:

The pros and cons of wide bandwidth, DC to MHz, versus narrow bandwidth, 20 Hz to 20 kHz, have been with us for more years than I care to remember. The controversy simmers and boils. Now it has come to a boil again.

J. Peter Moncrieff in his belatedly issued International Audio Review, July 1976, page 38, states that he considers a low-frequency cutoff of 0.5 Hz to be barely low enough for achieving accurate bass reproduction of music. On page 180 he considers an audio component with less than 0.5 MHz bandwidth to be audibly objectionable.

Mitchell Cotter believes that any bandwidth in a power amplifier (why not a preamp also?) in excess of 18 Hz to 35 kHz is detrimental to the reproduction of music and is asking you, as you say, to plunk down $300 to prove he is right.

This is a very serious issue to the dedicated audiophile interested in State of the Art equipment and should be explored thoroughly by The Audio Critic.

I am on the narrow bandwidth side even without the time compensation of the NF-1 filter. It has yet to be proven that phase distortion of the signal between the input and output of an amplifier is audible.

As you know, the reason for negative feedback is to reduce distortion. The wide bandwidth that is a concomitant of this feedback may be an asset or a detriment, depending which side of the fence you are on. There is a new school of thought (of which you are a member) that claims the less feedback the better the sound. I am inclined to think that the reason is the reduction in bandwidth of the less-or-no-feedback amplifier. With the NF-1 filter you can eat your cake and have it, too.

Think of the poor amplifier designer. What will a 20 Hz or 20 kHz square wave look like at the output of an 18 Hz to 35 kHz amplifier. Horrible, that's what. That it will sound better will not be believed by the reviewers who look at those square waves on their scopes.

Julius Futterman
Futterman Electronics Lab
New York, NY

*No comment necessary—except to remind all you young whippersnappers to listen respectfully when Julius Futterman talks. He didn't learn what he knows from a Tech HiFi salesman three weeks ago; he was designing state-of-the-art tube amplifiers when you guys were watching Howdy Doody.*

*—Ed.*

The Audio Critic:

When bi-(tri-, quad-, etc.) amping a system using electronic crossovers, the only speaker which needs a really high-slew-rate amplifier is the highest-frequency unit. In most cases, for high sound pressure level, low-distortion treble, a low-power high-quality amplifier can be used, such as Mark Levinson's new class A job. The rest of the power amps can be of lower slew rate, because the electronic filter networks will prevent any fast-rising waveforms from reaching their inputs. This is particularly true for subwoofer and woofer amps. High-slew-rate power amps with expensive, wide-bandwidth output stages are simply a waste of money for the low-frequency end of a system.

Since TIM and related distortions cease to be a problem for amplifiers driven by low-pass filters, large amounts of negative feedback and compensation can be used to get extremely low output impedance. This will create the high damping factors necessary for good, tight bass and midrange. Op amps such as the 741 and many of the low-frequency high-power output devices should be naturals for such amplifiers, bringing prices down drastically.

This should cut down the $20,000 price tag considerably on your Super Dooper Mark Levinson System.

Alan Hoover
Indianapolis, IN

*Cf. the letter from Chris Russell of the Bryston company, published alongside our power-amplifier survey in this issue.*

*—Ed.*

The Audio Critic:

Since I had not received the first two issues and have had to see them from other sources I am quite annoyed. It is quite obvious that you have a grudge against me. However, your lack of ethics are in question as far as I'm concerned. (*It are?—Ed.*) You accuse me yet allow me no defense. You compound this by refusing to honor my subscription which I have paid for. The enclosed photocopy is self-explanatory.

James Bongiorno
West Hollywood, CA

*Everybody is ganging up on poor Jim Bongiorno, including the United States Postal Service. His subscription had been sent to the GAS company and went astray; we sent him replacement copies to his home. The notion that we punitively withhold* paid *subscriptions from people we "have a grudge against" is downright pathological; of course, as Kissinger used to say to Nixon, "just because you're paranoid doesn't mean they're not out to get you." As for allowing no defense, any manufacturer with anything* factual *to say in defense of his equipment or his company has this column open to him, and we never said anything to the contrary.*

*—Ed.*

The Audio Critic:

It will be of some value to the other readers of your fine periodical if a couple of technical errors in the first two issues are corrected:

(1) In "Have Tone Arm Designers Forgotten Their High-School Geometry?" you criticized the Grace G-707 for its "incorrect" L sin $\beta$ of 3.375". (Did you *really* measure it accurate to .001"?) I have measured $\beta$ on my G-707 and have found it to be 23.25° (plus or minus 0.25°), which, in combination with the 9.33" L *specified by Grace* (L is adjustable by about 0.4", a fact you neglected to mention, which means therefore that L sin $\beta$ is adjustable) computes to, surprise, 3.68"; the exact number you described as being ideal. I think you owe Grace an apology.

(2) In "Fishing for Bass: A look at the Subwoofer Scene," you made the statement that a sealed system with Q = .707 has, among other things, "transient response . . . best possible for these conditions."

Although it is not clear what is meant by "these conditions," I can point out that, while a Q = .707 sealed box loudspeaker does have a maximally-flat (i.e. Butterworth) frequency-*amplitude* response, it does *not* have a maximally-flat (i.e. Bessel) frequency-*delay* response, and hence does not have the best transient response obtainable for a closed-box system. This occurs with a Q = .58. The difference between the two transient responses is that the former tends to ring just a little at a frequency equal to the cutoff (and resonant) frequency. In a subwoofer this is, of course, 20 to 30 Hz. Whether or not this is "audible" is an interesting but (as far as I know) unresolved question.

I would like, also, to contradict your somewhat pompous remark that there are probably no more than 15 engineers with a thorough grasp of the mathematics of speaker system analysis (*naturally* your boy is right in there around #7 or #9), since this writer has personal acquaintance with at least five. I suggest you get on with the business of reviewing audio equipment, and stop worrying about how incompetent everyone *else* is. Leave *that* burning topic to *The Absolute Sound.*

Having dispensed with my technical complaints, I have a more musically oriented complaint:

I wish you would find someone more empirically well-qualified to write about records and recording than Max Wilcox, who has produced the most consistently awful sounding classical discs ever produced in this country, and therefore the world. (I refer, of course to his Philadelphia/Ormandy recordings for RCA.) Why not Kenneth Wilkinson?

Sincerely yours,
Alan S. Watkins
Burroughs Corporation
Pasadena, CA

(*Six days later*):

In my (previous) letter I pointed out your measurement error regarding L sin $\beta$ of the Grace-707 tone arm, in "Have Tone Arm Designers Forgotten Their High School Geometry?"

Since that time, I have measured (directly) L and $\beta$ on the SME 3009 S2 Improved tone arm, and have found L (with Shure V-15 Type II) equal 9.14" and $\beta$ 23.0° (plus or minus .25°). This corresponds to L sin $\beta$ equal 3.57". which is within 3% of your ideal figure of 3.68": Allowing for 1) a 1% measurement error, 2) the fact that L is in small part determined by the cartridge used, and 3) that few of us are equipped with a laser to align our cartridges with our headshells, I think this geometry can be called optimum. This, in combination with the *exact* geometry of the G-707 referred to in my previous letter, would seem to show that your statement "We don't know of a single commercial design in which $\beta$ is optimum for the given L or vice versa," is based on your own faulty measurements.

Furthermore, I noticed that the latest issue of *Audio* contains a large ad placed by The Audio Critic, which makes use of the same misguided statement I have quoted above. This would suggest that the next candidate for roasting in "The Admonitor" is no one but The Audio Critic itself!

This kind of irresponsible sloppiness on your part is causing your credibility to deteriorate rapidly in my eyes, and I am sure in the eyes of other technically aware readers. We are not all asleep out here!

Sincerely yours,
Alan S. Watkins
Burroughs Corporation
Pasadena, CA

*It isn't our policy to publish and refute letters presenting technical arguments that are just plain wrong, "without redeeming social value," but this one is so aggressive and righteous in tone that we've decided to make an example of it. So:*

*1. As long as you don't twist the cartridge in the headshell (which is what we now recommend elsewhere in this issue but isn't what you're talking about), the L sin $\beta$ of* any *arm is constant and nonadjustable, even if L by itself is adjustable (i.e., if you can move the cartridge forwards or backwards, as in the Grace G-707). Why? Because when you change L, you're simultaneously changing $\beta$. But you aren't changing—and can't change without twisting—the L sin $\beta$. (Eureka! You ought to apologize to your high-school geometry teacher.) We've remeasured the G-707, by the way, and its L sin $\beta$ is nowhere near 3.68" without twisting.*

*2. For a fellow who's a little careless with his trig, you talk pretty good filter synthesis, and what you say about Butterworth vs. Bessel response is true—as far as you go. What you fail to mention is that a Bessel response profile has pretty grim amplitude characteristics, so that a Butterworth contour still represents the best trade-off between amplitude and time response —which is what we meant all along.*

*3. Leave it to a mathematical type to derive "no more than 15" from our "can probably be counted on two hands and maybe a foot." What's the matter, Alan? Don't they crack a few jokes around the water cooler at Burroughs?*

*4. Name a producer who has done better than Max Wilcox in that seventh-floor ballroom of the Scottish Rites Cathedral in Philadelphia. You think every recording site is the Concertgebouw?*

*5. How far off your SME arm is, if we accept your measurements as accurate, will be indicated by the fact that the optimum offset angle for an arm length of 9.14" is 23.75° and that the optimum arm length for an offset angle of 23.0° is 9.42". Since even fully optimized geometry in a pivoted tone arm yields a fairly high distortion index at the three maxima, we don't consider that kind of deviation to be negligible.*

*6. Maybe you aren't all asleep out there, but at least one of you is showing the effects of insomnia.*

*—Ed.*

The Audio Critic:

. . . One of your subscribers sent to me for comment part of an article by Bruce Zayde, titled "A Rational Approach to Low-Frequency Speaker Design", that part of the article in which he describes transmission lines as "nonoptimal designs . . ."

Since we manufacture damped reflex, infinite baffle, and transmission line designs, and find that the latter are quite superior to the former on spark gap and pulse criteria, we are rather at odds with Mr. Zayde, when he says that "a better approach to producing low bass response . . . is to stick with the sealed or vented direct-radiator format"; and also because we don't know of any such designs which in fact do provide "low bass," except transmission lines.

Perhaps Mr. Zayde would care to comment directly to us, on what is "better" about his recommended "format." I understand that your magazine features measurements, rather than purely subjective reactions. I am naturally interested in Mr. Zayde's measurements that corroborate his statements, not as a challenge to you, but based on my own curiosity; since most of the people around the world with whom I work would take my position, and base it on measurements.

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

*Mathematically, a transmission-line enclosure (or acoustical labyrinth) can be represented by a high-pass filter of the same order as certain types of vented (bass reflex) enclosures. Exactly the same expressions go into the mathematical model in each case, except that the transmission line has a large resistive component, corresponding to that long, heavily stuffed labyrinth. The upshot of this is that, by proper juggling of the interacting elements of either filter, you can come out in the same place, i.e., achieve exactly the same frequency and time response. And that means exactly the same sound quality. Mother Nature plays no favorites between networks of the same type and order. (We didn't have to consult Bruce Zayde to assert this simple truth.)*

*The reason why the transmission-line approach is "nonoptimal" is that the resistive component dissipates a lot of amplifier power; in other words, all other things being equal, the resulting speaker will be considerably less efficient, even if not lower in fidelity. The whole question was quite neatly disposed of by W.J. J. Hoge in his article on transmission-line speakers in the August 1977 issue of* Audio*, and there's an even more profound analysis in a 1975 thesis by G.S. Letts, a grad-*

*uate student of Dr. Small at the University of Sidney. When you get right down to it, the transmission-line bass enclosure has no serious academic backing; its support appears to be strictly by commercial and hobbyist interests.*

*When it comes to measurements, the question is—what exactly has been measured? Totally optimized executions of both approaches? We suspect that one reason why the transmission-line enclosure often comes out ahead in typical "as is" testing is that, by its very nature, it requires heavy bracing to be at all usable. Makers of ordinary vented (as well as sealed) enclosures can get away with less conscientious bracing, but in that case the performance isn't optimal. Our value judgment was based on what is possible in the light of available knowledge, not on what the industry is actually doing in response to audiophile mystique.*

*—Ed.*

* * *

*The rest of this column is devoted to the Mark Davis correspondence (in small type, as already explained). Enclosed with the letter writen to* ***The Audio Critic*** *was a copy of a previous (but until now unpublished) letter to audio journalist Randy Tomlinson, which should be read first for full understanding of the Mark Davis syndrome, and whose text follows.*

Dear Mr. Tomlinson:

Some work I had done with an undergraduate named Jay Gurley on the audible differences in phono preamps was discussed on the WBUR program "Shoptalk," and apparently it was a tape of this show that you heard and commented on, indicating why you did not believe our results.

The fact that people such as yourself should disbelieve our conclusions comes as no surprise. Indeed, I did not set out to prove that overall frequency response was the only important parameter contributing to preamplifier sound quality. I firmly believed, as you apparently still do, that two preamps could have identical frequency responses, as measured from a test record with a particular cartridge, plus levels of noise and distortion below audibility, yet still sound different. The explanation most people seemed to give for this had something to do with the high-frequency performance of the preamp, where "high" means outside the audio band: 30 kHz, 50 kHz, or more. Unfortunately, none of the explanations got much more specific than that and, although test results were quoted showing measured preamp differences in these frequency ranges, the fact remains that the ear cannot hear above 20 kHz, and that no record cutter in existence can cut frequencies higher than 18 kHz or so anyway. (*What about cutting at half speed, as some of the more sophisticated mastering studios do? Accepting your figure, that would give a high-frequency limit of 36 kHz. —Ed.*) Thus, what was missing was a clear-cut explanation of how the performance of a preamp at 50 kHz, and high levels of 50 kHz at that, could affect something in the audio band. What I set out to do was isolate that cause/effect relationship, not prove that it didn't exist.

Having built the zippiest preamp A-B box I could imagine, I connected my painstakingly-designed Davis-Brinton Reference Standard preamp (souped-up FET input, blazingly fast slew rate, precision this, that and the other thing, ultrasharp ultra- and infrasonic filters, triple-filtered power supply, et al.) and compared it to the worst preamp I could think of at the time: the two-transistor circuit copied almost verbatim from the GE transistor manual (which actually has three transistors; the output is buffered through an emitter follower, but why quibble) and found I could not tell them apart.

And this was after I had been a party to a couple of "informal" A-B comparisons, where we listened to one preamp for 20 minutes, then disconnected it, connected another, set levels to be vaguely the same, and found differences that were day and night. And which seemed to be repeatable.

After that initial A-B, Jay and I spent months trying to obtain the effects others seemed to be able to attain so effortlessly. We tried substituting every link in the chain: cartridges, turntables, power amps, A-B boxes, speakers, headphones, yet the results kept coming out the same: whenever there was the most barely discernible difference, there was always a corresponding difference in frequency response. Whenever there was no measurable difference in frequency response, subjects could not differentiate audible quality. Nor did I limit the tests to naive subjects: some of Boston's goldenest ears were enticed down to the lab. (And, incidentally, I don't believe in the idea of a person with golden ears: all persons with normal hearing have perception thresholds which fall within certain known limits. A properly designed psychophysical experiment should yield identical results whether the subject has a $3000 stereo and goes to concerts or not. However, this gets into areas of things like bias, training and feedback, which I don't want to try to explore here.)

Having failed to find anything relevant other than frequency response on our own, we approached some of the most vocal proponents of the other side, people like Al Foster of the BAS and Tom Holman of Advent. The listening tests they had done turned out to be informal to an extreme—which is not to knock them; they are not and do not claim to be psychoacousticians—but when we tried to isolate differences between preamps they claimed sounded different, only frequency response again seemed to be the operative parameter. Neither we nor they could hear differences when frequency responses were equalized.

(Of course, this put Holman and Advent in a sticky position, since the success of the 300 receiver was and is predicated in large part on the preamp section. Since our tests, Advent has been extremely careful with the wording of their ads; they don't claim the preamp is the best, only as good as the best. The fact that it's also as good as the $160 Pioneer SX-450 (discount price) and a host of others is something they fail to point out. They have, however, deleted all reference to the square wave test Tom talked about in the original AES preprint paper.

(This is actually a problem with the entire industry: conflict of interest. If it turns out you can do a preamp as good as any with 2 transistors, and the public learns of this, folks like AGI, Levinson, and lots of others have a problem. So do retailers who depend on high-end sales. So do equipment review magazines, who need something to talk about each month . . . Hell, not even Pioneer wants everyone to buy their $200 receiver and ignore all their higher-priced ones. This concept extends to other components as well—the industry needs things like TIM, phase response, moving coil cartridges, etc. to stay financially healthy. Frankly, I disbelieve most of it—I think frequency response is 90% of the story. The other 10% is cartridge tracking and loudspeaker radiation pattern.)

(*Like dynamic range is zero percent?—Ed.*)

Along the way, it did occur to us that long-term listening might reveal differences that A-B tests did not. Quite honestly, I think that there are some careful experiments that could be done in this area, but we did not pursue them because there is in fact already a small mountain of evidence to support their validity. In the parlance of psychoacoustics, an A-B test is normally referred to as a two-interval two-alternative forced-choice experiment (2I2AFC). A 2AFC is simply an experiment where either stimulus A or stimulus B is presented on a given trial, and the subject indicates which he thought it was. In a 2I2AFC, both are presented, and the subject indicates whether the order or presentation was A-B or B-A. There are other types, such as recognition experiments, in which one of a number of stimuli is presented and the subject indicates which. These experiments have been used to study a wide variety of acoustical phenomena, such as loudness perception, localization, speech intelligibility, etc., and in all areas that I am aware of, the 2I2AFC experiment results in the greatest reported subject acuity.

Our experience using the 2I2AFC paradigm in the preamp project strongly supported the validity of its use, in that whenever there was a measureable difference that fell above accepted thresholds of audibility, our subjects were able to pick it out on an A-B test. Among the differences that were audible: level differences of a very few tenths of a dB; frequency response differences over as little as an octave or so amounting again to a few tenths of a dB. At one point we attempted to use a home-brewed power amp comprised of a pair of Sanken 50-watt amplifier modules. Listening to a sine wave through them compared to a Crown 150 power amp revealed the presence of numerous high harmonics. Observation of the output waveform showed crossover distortion from the Sankens when they were connected to a loudspeaker, which was not present with the Crown. Spectrum analysis of the output confirmed the presence of the harmonics in the output of the Sankens, the absence of same in the Crown. Along the way, someone brought us an old HK tube preamp which apparently was defective. A-B testing indicated unusually high distortion, again confirmed with spectrum analysis. At one point, without informing the subjects, I hooked up an A-B which simply reversed the phase of one of the channels. The subjects, used to listening to ultrafine differences between preamps, were almost knocked over by the magnitude of the difference. Yet it took them almost 5 minutes of guessing to figure out what I had done, despite the fact that they were well-heeled in both audio and electronics. To me, this strongly reinforces both the validity of the A-B test procedure, and the notion that even experienced listeners are notoriously inaccurate in inferring the cause of differences they perceive.

(Incidentally, we made no attempt to control when subjects should switch, or how long they should listen before responding. If a subject wished to hear a whole cut again before making a single response, we played it. Inter-switch intervals ranged from a couple of seconds to well over a minute. Subjects were not told what to listen for. Thus, it seems that there was an element of longer-term listening involved here—at least over a period of minutes.)

There was one more check on the possible effects of long-term listening that I made. Although it was an informal test, I found the results somewhat revealing. In a second attempt to make a flat-response/lousy-sounding preamp, I took a lowly 1458 dual 741 op amp, put an RIAA feedback loop around each half, and called it a preamp. With a pair of 9-volt batteries the whole thing fit into a small minibox. I A-B'd it against the other preamps in the lab and, after tweaking frequency responses and input capacitances, the 741 preamp was indistinguishable, except at very high listening levels with exceptionally quiet records, where the difference in noise level was apparent audibly. Even there, the difference was small: with a V-15 III cartridge connected, the preamps we looked at all had S/N ratios of about 82 dB, A-weighted, re 10 mV, 1 kHz. The 741 was 77 dB. After the A-B tests, I was still very suspicious of the results, so I took the 741 preamp home, where for months I had been listening to a good-sounding Davis-Brinton unit. My first reaction after substituting the 741 was that things didn't sound right, but I couldn't put my finger on the cause, so I left it in the system, in place of the Dav-Brit. An hour later, I looked up pleased with how well the system was sounding until I realized that I was listening to the 741 preamp; then it didn't sound quite right again.This phenomenon was repeated several times over the next day or so. As long as I didn't think about what preamp was on, the system sounded as good as ever, but as soon as I realized that the preamp consisted of nothing more than a 741, it stopped sounding good. Clearly my prejudice against the 741 was manifesting itself in very audible terms, whenever that prejudice was allowed to function. When the prejudice wasn't active, neither were the sonic deficiences. Eventually, after a couple of days, I guess I became convinced of the basic goodness of the 741 preamp, and I stopped "hearing" deficiencies. I loaned this preamp to Peter Mitchell, of the "Shoptalk" show, and he subsequently informed me that he thought it was one of the better sounding preamps he had heard.

This I believe illustrates one of the deficiencies I perceive in the tests you briefly describe in your letter. Namely that the people doing the long-term listening tests know what preamps they are listening to. (Or so it seems from your description.) I am afraid that this factor alone completely invalidates any results you might get. I maintain it is totally impossible to A-B knowingly a preamp made with $2 in Radio Shack parts and chicken wire to a $1200 preamp, without experiencing some measure of bias, and that this bias will manifest itself in apparent audible differences, even if the two circuits are in fact electrically identical. (*You think any two resistors or capacitors of the same rated value and tolerance are electrically identical? Not so. —Ed.*) In running our tests, the subject was never informed in advance which preamps he was listening to. Nor was any feedback given after each trial; if we are talking about differences that really are hearable, you shouldn't have to tell the subject when he's right or wrong, he should tell you.

(*A lengthy passage discussing the proper way to match cartridge/preamp frequency responses for A-B testing is deleted here, since the main issue has little to do with laboratory technique. —Ed.*)

There are a few other points in your letter that I would like to comment on. You take note of "certain superior characteristics of the JC-2 (imaging, definition, clarity)." While these terms may have some meaning to you, and may be valid descriptions of what you think you hear, I know of no serious psychophysical work that has been done to show these as being real, repeatable, measurable effects. The closest I can come is the concept of image fusion, from studies in localization and binaural lateralization, which is a function of how well the two channels are matched. The characteristics and thresholds of that matching have been extensively studied, and there is no preamp that I know of that comes within light years of having sufficient channel mismatch to cause detectable deterioration in binaural fusion, especially when auditioned over loudspeakers. (*Never heard of cross talk, eh?—Ed.*) The channel mismatches present in cartridges, records, loudspeakers, rooms, and ears far exceed those of preamps. Frankly, I suspect qualities like imaging, definition, and clarity are strictly functions of frequency response, level, and, alas, imagination. As you note in your letter, "the main problem is the psychological phenomenon that occurs in our heads. We seem totally unable to control the way the differences grow as exposure time increases." I agree with that statement in part, although I think there is a limit to how far those differences can grow. The problem is that a subject will tend to reinforce a perceived difference whether or not it exists. Suppose for example that when switching from switch position A to switch position B one time, the high frequency content of the program material drops. The subject might begin to suspect that the preamp in the A position was brighter than the B preamp. Instead of asking himself, "Is there a difference between A and B?" he will now be asking himself, "Is A brighter than B?". The latter question shows bias, the former does not. Faced with a psychological compulsion to find a difference, the subject will tend to reinforce the perceived difference, real or imagined. If, however, the correspondence between switch positions and preamps is switched randomly after each trial, the subject can't always be sure that A will be the brighter one. Thus if the difference was only imagined, the subject's responses will be guesses, and he will be right only 50% of the time, providing enough trials are done. The point is that this psychological reinforcement makes it imperative that the subject not know what he is listening to, and that enough random trials are carried out to account for guessing, even in long term listening tests.

When we started out to study preamps, we realized it would be impossible to try every available preamp, power amp, turntable, cartridge, etc., in every combination, so we chose what we felt was a representative sample of equipment, making sure that everything was duplicated at least once, and that the preamps covered as wide a range as possible. Nevertheless, it comes as no surprise to have someone claim that the reason we couldn't hear differences was that we didn't use thus and such a cartridge, arm, power amp, speaker, whatever. Frankly, I don't find the argument terribly compelling. In the first place, the experts out there can't seem to come to any consensus as to what the finest equipment is. You claim we should have used a Dyna 400 or an Ampzilla, but I seem to remember reading some other review that said the Phase 400 was the hot setup. Then I think there was another one that said that the BGW, which we did try, was the only way to fly. In the second place, nobody ever gives a solid cause-effect reason why their particular super product is alleged to actually sound better than the other guy's. (You say your preamp has .0008% distortion? Why should that sound better than their .01% preamp when the threshold of audibility is .2% under the best conditions, and when the cartridge is producing 4%?) Not even Mark Levinson can tell you why his preamp is supposed to sound better in explicit terms.

In the third place, as I've mentioned, whenever there was a certifiable difference, in level, frequency response, distortion or phase, the subjects heard it—and heard it as soon as it exceeded recognized thresholds of audibility. On all of the test setups.

(*But earlier in this same letter you bracketed phase response with industry hypes like TIM and moving-coil cartridges.—Ed.*)

In short, Mr. Tomlinson, you seem to be asking me to believe in magic. You claim there are preamps out there with conventional specs—things like frequency response, distortion, noise, input impedance—identical to the "finest" preamps, which nevertheless are audibly inferior, but you do not tell me what these inferior preamps are doing to the signal to screw it up in an audible fashion. You claim the audible differences start out as barely audible in A-B tests, but become more audible with casual long-term listening, without explaining why. You claim these differences are audible only on "state of the art" equipment, without providing an explicit definition of what makes a given piece of equipment state of the art. Having failed to specify the magical ether that seems to make it through the good preamps, and not the bad, you are obviously not prepared to say how it gets through "good" amps, tone arms, cartridges, and speakers to become imaging, definition, clarity, what have you, upon striking our ears.

About 12 months ago I set out to try to find that ether. I have been trying assiduously to track it down, but somehow it always seems to hide behind a tree when it sees me coming. I'm beginning to think it unfair that I should be the only one to have to carry the burden of proof. (*Very interesting. Now tell the doctor when you first began to experience this feeling of isolation.—Ed.*) Before too many more people are coerced into blowing twelve hundred bucks on mystique, I think it would be nice if someone showed in clear, irrefutable repeatable terms exactly what those twelve big ones buy in audio quality, and how that can't be just as easily bought for $17.50. In other words show, using controlled, blind listening tests, short-term or long, that a difference really exists, and then provide a clear cause-effect relationship that explains the origin and behavior of those differences. Then justify why those differences should cost as much as they seem to.

After some record producer has tossed equalizers, filters, limiters, compressors and reverb into the signal chain with gay abandon, I don't see why I should spend 5 times more for a preamp just to get an RIAA response that's 0.2 dB more accurate at 20 kHz. At the very least, if there are differences that can only be discerned on "near state-of-the-art" equipment, I think your readers should be so informed, so they won't waste large sums of money on a preamp if they happen to own AR-11 loudspeakers, or other equipment you deem not near state of the art.

I am willing to have my claims proved incorrect, Mr. Tomlinson, but I don't believe you have done so.

*And now for the Mark Davis letter writ-*

*ten directly to us, in response to our preamp survey.*

The Audio Critic:

I was recently given the opportunity to read an article published in your premiere issue which discussed at modest length certain research concerning RIAA phonograph preamplifiers to which I was a party. The principal finding of this research was that competently designed phonograph preamplifiers, ranging in complexity from the costliest, most abstruse topology, to the simple and popular two-transistor configuration, are in fact audibly indistinguishable under any conditions of normal operation (i.e. reproduction of commercially available records by conventional magnetic pickups), providing the concept of competence in design subsumes the equivalence of overall playback frequency response to a tolerance of 0.3 dB or better. (*Moving-coil pickups are also magnetic, we subsume?—Ed.*)

It is possible that certain misconceptions may have been promulgated by my brief discussion of the research on the WBUR radio program, "Shoptalk," which you indicate as the source of your information. To the extent that you find it enlightening, I am enclosing a copy of a letter I have previously sent to Randy Tomlinson, of Accusound Engineering and *StereOpus* magazine, which treats in somewhat greater detail some of the questions at issue.

There are a few other questions your article raises that I would like to attempt to clarify. These generally involve electrical measurements or listening technique.

I concur with your conclusion that neither harmonic nor IM distortion is a significant factor in modern preamplifiers. Sometimes (too often) the opposite conclusion is reached by driving the preamplifier being evaluated with electrically generated test signals to levels hopelessly greater than will actually be encountered by the preamplifier under actual conditions of use (cf. Jung, 1977).

Obviously, I also take no exception to any reservations you may have about tests employing square waves, wideband noise, or other similar, unrealistically fast test signals. To the best of my knowledge, Tomlinson Holman no longer believes his square wave test to be relevant; nor does Alvin Foster endorse his white noise test (Foster and Swanbon, 1977).

It is gratifying that you find a preamplifier with relatively low slew rate (1 volt per microsecond) can deliver excellent sound. Our calculations indicate that such a figure is at least an order of magnitude larger than what is required by available records and cartridges. An interesting sidelight is that slew rate figures comparable to 1 V/uS are obtained from typical well-designed preamplifiers employing the two-transistor topology.

Your results on signal-to-noise ratio, that some units were "measurably and audibly quieter than others," and that some were "outstanding" (p. 8), runs somewhat counter to our own measurements, which showed fairly strong consistency among the units tested. This may be due to differences in measurement technique; regrettably, you do not discuss the protocol you employed.

As you may know, the noise of a preamplifier is lowest when the input is terminated with a short circuit. In trying to present the most attractive-looking figures, manufacturers will often specify the signal-to-noise ratio of their preamplifiers under this constraint. Occasionally, one will even optimize for it, to the detriment of the conditions of actual use; i.e. with the input terminated by a magnetic cartridge. Hallgren (1975) has shown that the principal source of noise in a preamplifier is in fact the cartridge, followed in importance by the 47K ohm input termination resistance. Since it is the normal mode of operation of the preamplifier, terminating the input with a cartridge while measuring its noise seems to us to be the most realistic approach. True, the result will depend to some degree on the cartridge chosen; nevertheless it will be more representative than the shorted input specification. The latter is probably more apropos when a low impedance device, such as a pre-preamplifier, is connected to the preamplifier. Even then, it is of secondary interest, since the cartridge and pre-preamplifier input noise will inevitably dominate, to be masked in turn by record noise. The need to terminate the input with a cartridge applies as well to the measurement of the input noise of a preamplifier, incidentally.

We used a Shure M91E cartridge; while its sonic qualities may not be state of the art, its inductance and other electrical characteristics are typical of a large number of current state-of-the-art cartridges. With the cartridge mounted in a shielded box to suppress stray hum fields, and the stylus removed to avoid acoustic pickup, we found almost all preamplifiers had a signal-to-noise ratio of 82 dB, ASA 'A' weighted, re: 10 mV RMS, 1000 Hz. This is in good agreement with Hallgren's data.

We checked these measurements by listening to the noise of the preamplifiers, after first taking care to match levels with a test tone, with the respective inputs terminated in turn with the same cartridge. While there were clear variations in the spectrum of the noise among units examined, the overall level remained essentially constant.

Basically, the same laws of solid-state physics and thermodynamics hold for all hi-fi companies. For about 10 cents you can buy, in production quantities, a transistor that will perform as well as possible with available cartridges. While there are some games you can play with the 47K input resistance, only small gains beyond 82 dB are possible, and even the least expensive receivers appear to hit that figure routinely. We have thus concluded that the concept of an ultra-quiet phono preamplifier is largely a myth. *(What about small-ly? That's our bag.—Ed.)*

One of the largest areas of disparity among preamplifiers was found to be input impedance, so I was a little sorry to see that you failed to measure it. The use of a low-inductance cartridge probably minimized the effect in your tests, but careful measurement of relative frequency response, including the effects of the cartridge, as described in the accompanying letter, would have been preferable to assuming the input impedance had no effect. Furthermore, not everyone uses the cartridge you employed; many popular cartridges are sensitive to variations in input impedance, raising the possibility that a preamplifier that sounded fine in your tests would deliver inferior sound with a cartridge having greater inductance; ergo the importance of reporting this specification.

(*We don't know of a single high-inductance, i.e., input-capacitance-sensitive, cartridge that can compete sonically, regardless of loading, with moving-coils like the Denon or EMT, or low-inductance moving-fields like the Grado. We don't recognize the "rights" of low-fidelity cartridges; maybe that's why we hear preamps differently, too.—Ed.*)

Of greater concern to the accuracy of your report than the results of the electrical tests are the psychophysical issues implicit in your listening protocol.

Your measurements of preamplifier frequency response show unit-to-unit variations on the order of 0.75 dB, a tolerance which you claim is "good enough to eliminate the possibility of listening preferences on that score" (p. 7), further commenting that requirements of finer tolerance are to be considered "obsessions" (p. 8). As noted above, our investigation indicates the need to equalize overall playback frequency response within a tolerance of 0.3 dB. That is, the cumulative effects of input impedance, frequency response, and playback level must not induce a difference greater than 0.3 dB in level between the two preamplifiers being tested, at any frequency in the audio band. Preferably, the difference in overall level, weighted on an octave basis, should be zero.

This constraint was not imposed capriciously; it resulted from concrete psychoacoustical data. Using the same AR speakers you think so little of, and the same A-B paradigm you denigrate without quantitative justification, we found subjects could reliably discriminate between sources solely on the basis of frequency response disparities of as little as 0.7 dB. By limiting differences to a maximum of 0.3 dB, frequency response/level discriminability was found to be virtually eliminated, permitting accurate acoustical evaluation of other possible disparities. The tolerance of $\pm$ 0.75 dB you encountered implies a worst-case deviation of 1.5 dB, which, when added to the tolerance of the level match you made, 1.0 dB, results in a maximum possible deviation of 2.5 dB, not counting the possible effects of input impedance variation.

Your plausibility arguments on pages 7, 8, 11 and 12 notwithstanding, this difference is, by actual empirical observation, well beyond the point of becoming plainly audible.

Your failure to carefully measure and control playback frequency response/levels (casual tweaking with tone controls does not constitute careful measurement and compensation) makes it possible, nay likely, that your judgments primarily reflect preferences in frequency response. Of course, it is our contention that this must be so, as there is no other audible variation in preamplifiers; but even if there is, it is likely that it was swamped out by the potentially large variations in frequency response and levels.

You dismiss the AR loudspeakers as "too forgiving" (p. 10), without citing any references to back up this claim, or stating how one goes about measuring the "forgivingness" of a loudspeaker, or what specifications contribute to this alleged quality. (*References? If authoritative criticism of speaker systems—by brand name!—were available in references, we wouldn't be in business. —Ed.*) We acknowledge that there are loudspeakers which have flatter frequency response than the ones we used, but this is not a necessary prerequisite, since the same speaker is used for both A and B presentations. What is important is sufficiently wide response, without serious gaps, to ensure that all audible frequencies are reproduced at hearable levels, plus sufficiently low distortion to avoid masking possible distortion from the preamplifiers being auditioned. (*But what kind of distortion? Steady-state or time-dispersive?—Ed.*)

Both of these criteria are met by the AR speakers. We have measured the IM distortion of a number of commercial tweeters and found it in all cases to be under 0.5% at a level of 100 dB SPL, 6 feet on axis, the distortion dropping rapidly below this intensity. This is less distortion than the ear itself produces at such levels. (*Yup. You meant steady-state. —Ed.*)

The most relevant justification of the speakers and associated equipment we used was that subjects were able to clearly discriminate between sources whenever there was a certifiable difference exceeding accepted thresholds of audibility, as for example in the case of frequency response/level deviations of only 0.7 dB (which you believed to be inaudible). This fact, plus the strong consistency in results with alterations in experimental equipment, including the substitution of Koss ESP-9 electrostatic headphones for the loudspeakers, lends fairly strong empirical credibility to the results we obtained.

(*Hold everything! We never said that differences of 0.7 dB are inaudible, but rather that they aren't relevant to our conclusions. As for the electrostatic headphones, did you use them when you A-B-ed the two-transistor GE preamp against the Mark Levinson? Right then and there? —Ed.*)

Lacking a proven alternative mechanism to account for differences in preamplifier sonic quality, your allegation that our equipment fails to preserve the effects of this unknown mechanism would seem somewhat premature at best.

The rationalization you advance for eschewing the use of an A-B test is that it fails to show up differences that are really there, because "your ear tends to integrate the sound under these conditions" (p. 11). This position is simply untenable; there is no such integration effect to mask differences.

As explained in this letter and the one accompanying it, our A-B tests revealed extremely fine differences in level, frequency response, and distortion. Beyond that, numerous diverse disciplines in psychophysics make use of the instantaneous two-interval A-B discriminability paradigm, usually to explore the highest levels of human resolution.

I know of no authoritative, controlled study showing long-term listening as you describe to be a more accurate indicator than A-B testing; if you know of one, please cite it. In the meantime, I commend to your attention the *Handbook of Mathematical Psychology* (1963) as a viable primer on the subject of competently conducted listening tests and their analysis.

The attitude toward statistical testing is perhaps one of the most contradictory elements of your article. You claim, without providing justification, to have "very little faith in statistical surveys ('19 persons preferred A and 22 preferred B')" (p. 10), implying incorrectly that a preferential methodology was employed in our tests. Having made this claim, you then execute an about-face, and perform just such a preferential survey.

Frankly, I completely agree with your lack of faith in such surveys, unless the basis for listener preferences can be objectively isolated, and unless it can further be shown that the preferences expressed represent a fairly universal consensus. Otherwise, it is indeed true that one may "come to a different conclusion with a different selection of listeners" (p. 10). You have made no attempt to show quantitatively that your preferences would remain the same with a different, independent group of listeners. Indeed, you have not even shown that your preferences are independent of the equipment chosen; to the contrary, the single substitution you did make, of the power amplifier, reversed the order of preference of the highest ranked preamplifiers.

(*That's right. We were trying to help our subscribers choose a good preamp, not to write a textbook on psychoacoustics. —Ed.*)

Almost without exception, the human auditory system responds in a stochastic fashion; present a stimulus to a subject at two different times and, in general, you will get two different responses. Such behavior makes the need for accurate statistical characterization manifest.

For the record, we never asked a subject which preamplifier was preferred, only whether there was a discernible difference. Subjects were given no feedback or prior information that could cause bias; the sound of the preamplifiers was the only information available to the subject on which to base a decision.

The probabilistic nature of the results is reflected in the fact that most people could discriminate frequency response disparities of several dB most of the time; some people could discriminate frequency response disparities on the order of a major fraction of a dB some of the time; and no one could reliably discriminate disparities of 0.3 dB or less. Allegiance on the part of the subjects to a particular concert hall or type of music had no effect on the results. Toole (1977) has recently presented some data indicating that experienced listeners ("golden ears") are not even required in preferential studies; naive listeners can be just as consistent.

In summary, your listening tests incorporated some very serious lapses of accepted psychoacoustical procedure: the failure to equalize for differences in level and frequency response; the failure to keep subjects from comparing opinions, despite the need for statistical independence; and the failure to conduct independent blind trials, without feedback, prior information, or other sources of bias, in sufficient number to engender reliable statistics.

Even long-term listening tests, for which you express preference, can be done properly, providing the same rules of technique are observed. As it stands, I doubt that any competent psychophysicist would endorse your methods or accept your conclusions.

Lurking behind these issues is the conceptual question of whether it makes any sense to test preamplifiers when you profess not to know what makes them sound different. For one thing, the perceived superiority may not be worth the extra cost of the better units; it may be possible to incorporate the extra "goodness factor" which you claim the better preamplifiers possess into a cheap preamplifier for next to nothing. For example, if you assume for a moment that our contention is correct, that frequency response is the only significant parameter, then it is foolish to recommend an expensive unit when an inexpensive one will either have the same response, or can be adjusted to have it by a suitable choice of input capacitance.

Without knowing the specific mechanism(s) involved, you cannot be sure that your conclusions are true in general for other installations. Again, as an example, if frequency response is all important, a revised choice of associated reproduction components would very likely have altered the preferences. As already noted, switching just the power amplifiers changed the order of the two best-ranked preamplifiers. Without an alternative mechanism, clearly identified and characterized, you cannot prove that your preferences will hold for anybody else.

It is significant that the results and conclusions we reached have also been obtained by the "highly active and enlightened" (p. 9—here, we agree completely) Alvin Foster, of the Boston Audio Society, who formerly espoused the relevance and use of tests employing very high-frequency signals, such as wideband noise. Some of these results will be published shortly (Foster and Swanbon, 1977).

The same conclusion was also reported to me by David Griesinger (1976), of Harvard University and the Boston chapter of the Audio Engineering Society. The phenomenon thus does seem repeatable.

Hopefully, this information will help clear up some of the misunderstanding that seems to exist. I know from personal experience that the results of informal listening tests often seem so obvious as to make a more precisely controlled environment appear unnecessary. More often than not, however, the impression is a false one. Before rejecting accepted psychophysical experimental protocol, you are urged to try it.

Sincerely,
Mark F. Davis
Massachusetts Institute of Technology
Cambridge, MA


References:

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*Our overall reaction to these two letters is that, even though they fly in the face of all the slowly and painstakingly accumulated wisdom of an entire generation of audio perfectionists, they mustn't be dismissed out of hand. Obviously Mark Davis and his associates are highly knowledgeable, dedicated and sincere. What they claim they heard is undoubtedly what they actually heard—and quite possibly, had we been there, we too would have heard the same thing. The question to ask is why it sounded that way.*

*The most plausible explanation is that the system into which the preamps under test were inserted had insufficient resolving power. Our experience is that plugging different preamps into, say, a Beveridge Model 2SW system makes the differences laughably obvious, regardless of matched listening levels or RIAA equalization accuracy. (The signal source being a Denon DL-103S cartridge, fanatically aligned for optimum tracking laterally and vertically.) Through Dahlquist DQ-10's, used in our original survey, the differences are still easily audible though less flabbergasting. Through the Mark Davis setup—who knows? We have a feeling that total disregard for the masking effects of time-dispersive distortion—originating from the cartridge and the speakers—and exclusive concentration on amplitude response and steady-state distortion channeled the Mark Davis tests in a self-fulfilling direction.*

*We say self-fulfilling because we also have the distinct impression that not only Mark Davis but the entire Boston Audio Society crowd (delightful and sophisticated practitioners as many of them are) still would like to discover an audio Nirvana for forty-nine dollars and ninety-five cents. We discern in them a peevish hostility toward ultrahigh-end equipment, perhaps because so many of them are academics and empathize with music-loving students who can't afford a Mark Levinson ML-1. And, of course, the best way to dismiss the expensive stuff is to find it scientifically indistinguishable from the run of the mill.*

*We must also remind our readers, to put things in proper perspective, that Mark Davis also considers TIM in power amplifiers to be a myth. Get the picture? It's not only the emperor who is naked, but also the empress, the crown prince, the archduchess . . .*

*—Ed.*

# The Realities of Noncommercial Audio Journalism

By Peter Aczel
Editor and Publisher

**As we go into our fourth issue, we realize that some of our subscribers still don't perceive us as we really are and don't quite know what to expect of us. Hence this continued exposition of our philosophy and of the facts of life in our delirious business.**

Although what follows is only loosely connected with the two previous editorial discussions of our point of view, we prefer to continue the sequential numbering of topics for convenient reference.

* * *

**16** Just because our publication doesn't make a totally amateurish impression in looks and contents, please don't assume that the publisher is McGraw-Hill or some other such multimillion-dollar operation. We despair when, after having tested and reviewed more than 30 preamplifiers and preamp accessories in a little over four months, we get letters saying, "I'm shocked that you failed to include the following 11 models." How big and efficient do these letter writers imagine we are?

The fact is that we're about the same size, in available space and personnel, as other audio reviews that don't carry manufacturers' advertising. What distinguishes us from the others is that (a) we work harder—or at least generate more output in a given time, (b) we were willing and able to make an initial capital investment in a completely equipped, in-house laboratory facility, and (c) we try to correlate what our ears tell us with the laws of physics and mathematics rather than techno-folklore or wishful fantasies. None of these distinctions enable us, however, to test 600 pieces of equipment in a year or even to meet deadlines with unfailing punctuality.

Our basic limitation is, of course, economic, and it's intrinsic to our noncommercial format. Let's say we decided to hire a full-time staff engineer with a reasonably sophisticated audio background (rather than using part-time staff consultants as we do now). His salary would wipe out the proceeds of about one thousand new subscriptions, right up front. An experienced full-time managing editor would wipe out another thousand. How many thousand subscribers do you think we have? Now if *Audio* magazine wanted to hire the same engineer, they could do it with the income from one full-page ad per month. See the difference? There, in a nutshell, is also the reason why we can't sell **The Audio Critic** at a dollar a copy. Or even two dollars.

* * *

**17** You must remember that the original concept behind noncommercial audio reviews was that testing exclusively by listening can be highly revealing in a relatively short period of time and requires no test instruments other than educated ears. Therefore the operation can be started with minimal manpower and overhead, with sufficient income from subscriptions and possibly dealer advertising to maintain a reasonable publishing schedule.

Now here's the paradox: **The Audio Critic**'s way of testing is, on the face of it, incomparably more time-consuming than anyone else's, since we keep hopping back and forth from listening room to laboratory, trying to find a correspondence between what we hear and what we measure. This also dissipates a much greater percentage of subscription income on overhead before editorial and publishing expenses. By all rights we should be the slowest, most plodding noncommercial audio journal of them all. The fact that we aren't, that we cover more ground more thoroughly in a shorter time than the strictly golden-ear boys, is one of the best arguments for parallel golden-ear and laboratory testing. Because, in the long run, the scientific (or call it technical) approach actually saves more time and energy than it uses up.

We could spend weeks trying to determine purely by listening tests whether an elusive low-frequency resonance we hear in a speaker is inherent in the design itself or is caused by some subtle interaction with the room, the walls, the furniture, the rest of the system, or whatever. Five minutes of spectrum analysis with the proper frequency span and resolution bandwidth will, on the other hand, provide an unequivocal answer. Occasionally it may not even be necessary to perform a laboratory test to save listening time; simple inspection by someone who understands the applicable design principles will suffice. For example, a tone arm with wobbly bearings can't possibly sound as good as one with tight, preloaded bearings (other things being equal). It's an obscene waste of time to let several keen-eared people listen to such an arm for weeks and weeks only to have them report that there's something not quite right with the sound. So, an equipment reviewer who is into bearings as well as Mozart and Pink Floyd can confidently say "Forget it!" ten minutes after unwrapping the arm, instead of staging an elaborate blind A-B test against a reference standard.

We recently gave a somewhat oversimplified summary of this point of view to someone who asked us how we perceived the difference between our procedures and those of another noncommercial audio review. Suppose, we said, that a manufacturer came out with a turntable he claims to be a perpetual motion machine. The other publication would say, "A perpetual motion machine? Let's have a look." We'd say, "A perpetual motion machine? Get the hell out of here." We have no time for audio designers who defy the laws of nature.

* * *

**18** Our special CES issue (Volume 1, Number 3), in which several such scofflaws were pointed out and dismissed out of hand, provides another illustration of how our departures from the expected practice of noncommercial audio publications can be misconstrued by certain subscribers. We published this issue with the firm conviction that it filled an important need (way in advance of commercial magazine coverage!) and was fully up to the standard set by our previous two issues. We felt that, quite aside from their news value, our CES commentaries shed some interesting bits of light on a number of subjects ordinarily shrouded in mythology, such as stylus shape, tone-arm geometry, woofer design, the bandwidth-limiting controversy, and others. We were enthusiastically praised for this by more than a few distinguished members of the audio community, not to mention advanced audiophiles. Nevertheless, about 1% of our subscribers turned out to be extremely unhappy about the whole thing. (To be precise, 1.4% communicated with us specifically in regard to the change of pace set by the third issue: 0.4% pro, 1% con, 98.6% silent and presumably unruffled.)

It was our distinct impression that the majority of the complainers didn't really dig into the issue but took one quick bite, found that the flavor was different than last time and started to holler. They certainly seemed to be unaware of the aforesaid little raisins and chocolate chips inside. Sorry, guys, we never said it would be vanilla every time. Nor did we ever promise 48 or 52 or any other number of pages per issue for your $4.67. We use up as many pages as we feel is necessary to cover the subject. In the future, there will undoubtedly be 64-page issues as well as 24-page issues. We sell information and advice, not yard goods.

To keep everybody happy, however, we don't plan to publish a separate CES issue in 1978 but will probably cover the CES in one of our long surveys within the framework of a normal (i.e., fatter) issue.

* * *

**19** The subscriber who gets belligerent about the number of pages per dollar, the number of words per page (our type size is large, see?) and that sort of thing seems to be the same one—representative of a very small minority, mind you—who would like us to run long laundry lists of recommendations without any fancy guff about slew rate or sidebands or Q. Preferably twice a month. (Mark Levinson ML-1—good, Dynaco PAT-5—bad, Dahlquist DQ-10—good, Janis W-1—bad . . . Next!) We hate to disappoint such seekers of predigested simplicity, but that's not our idea of a serious audio journal. We'd much rather coax, cajole, browbeat, lecture and just plain educate our subscribers (at least those that aren't hardened professionals) to the point where they'll trust their own judgment, even about products they've never seen reviewed. If we can accomplish that, the difference between 52 and 24 pages, or $4.67 and $3.50 per issue, will begin to appear rather irrelevant.

And when that day comes, we'll get no more letters saying, "I just can't wait till you review the Schmidlapp C-95—I've owned one for six months." The last thing an enlightened audio enthusiast should need is a review of a product he already knows intimately.

* * *

All right. We'll stop biting the hand that feeds us and present herewith what we believe is our best, most useful issue so far. Read it, and then see if the above comments have helped to clarify who we are and where we stand.

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***Editor's Note:*** *Consultation by telephone on individual purchasing decisions or installation problems emphatically isn't part of the services offered by* ***The Audio Critic*** *for the price of a subscription, even if you're resourceful enough to track down the Editor's home phone number.*

*"That was the last test, Otto.
The prototype is finished."*

*"All we need now is J.B.'s approval.
Wait till he hears this one!"*

*"J.B., you've just got to come down
to the lab and hear the new speaker.
It's fantastic!"*

*"Eh?"*

Calderhead

# Sophisticated Speaker Systems, Large and Small: A Comparative Survey.

By the Staff of
The Audio Critic

**Part I:** Comprising seven big ones, four little ones, three medium-sized jobs, plus a decent subwoofer, which should be enough to keep everyone arguing until Part II.

There are good reasons why speakers are the most controversial of audio components. First of all, they are the least accurate, introducing much greater discrepancies between input and output than amplifiers or tape recorders or even phono cartridges. That makes most comparisons highly dilemmatic, as they are essentially between varying degrees of "bad." (Who was the nicer guy, Adolf Eichmann or Charles Manson? You're wrong!) This basic judgmental problem never goes away; it's there whether one is comparing the Beveridge and Koss electrostatics or the house-brand specials of a hi-fi chain store. Name your preference and someone will immediately tell you what's wrong with it. And he'll be right. No speaker, at any price, is without faults.

Another cause of unending controversy is that a lot of audio enthusiasts with strong opinions have never heard, let alone lived with, the best available speaker systems, which are usually made in extremely limited quantities and demonstrated in very few places. As a result, the typical arguments tend to be parochial, as they are engaged in without awareness of what is currently *possible*. Amazingly, even speaker manufacturers suffer from this limitation; you'd expect their sound rooms to bulge with the latest and greatest multi-kilobuck speakers for reference purposes, but the truth is that most of them (rich or poor!) listen exclusively to their own products. Their dealers are usually way ahead of them in listening experience.

The most profound source of confusion, however, is the quality of the signal fed to the speakers in nearly all comparisons. A record made with peaky Neumann microphones is placed on a turntable with insufficient feedback isolation and a motor board that drums. A cartridge mounted conventionally, with unnecessarily large tracking error, in a tone arm with sloppy bearings (cf. our cartridge/arm/turntable article in this issue) is lowered on the record. The signal is applied to the speakers through an amplification system with TIM, insufficient voltage swing, power supply limitations, etc., etc. The speakers are then found "crisp" or "nasal" or "honky" or "airy" or whatever. And these observations are then passed on to audiophile friends or even published in underground audio reviews. If at least the signal had the *same* faults in each listening test! But, needless to say, its quality varies completely at random, and so do the conclusions.

**How we tested them.**

What we said about A-B listening tests in connection with subwoofers in our second issue (March/April 1977, page 30) applies equally to full-range speaker systems. The audible differences are so gross, between any two models at any volume level, that A-B switching becomes a pious exercise in methodology yielding no additional information. When you can find two really good speakers that have to be carefully A-B-ed to establish which sounds more accurate, you'll know the art has advanced by a whole order of magnitude.

On the other hand, one-by-one listening evaluations require absolutely unquestionable signal quality, and we went all out to obtain it. Only the finalists of our preamp and power amp surveys were used (incidentally, the new Hegeman HPR/CU preamp has emerged as a very serious contender for reference status, although we won't be ready to rate it against the Mark Levinson, Rappaport and others until the next issue); our current reference cartridge, the Denon DL-103S, was religiously aligned for optimum tracking both laterally and vertically; we eschewed all records, no matter how dear to our heart musically, unless they were obviously recorded with *flat* microphones, like the Mark Levinson Acoustic Recordings; and, to establish an ultimate reference point, we unleashed our new, highly modified Stellavox 'Stellamaster' tape deck, playing direct 15-IPS copies of selected original master tapes. (Please don't write us to ask how we buy, beg, borrow or steal these.) In other words, we were listening to the speakers, not to signal distortions ahead of the speaker terminals.

After extensive listening, we measured. We made a point of starting our laboratory tests only after we had formed a reasonably firm opinion of how a speaker sounded and what was audibly right or wrong with its performance. We didn't want to expose ourselves to the temptation of subjectively aha-ing and you-seeing from our listening chair what we had already established objectively through measurement. Certainly not the first time we heard a speaker. All the initial aha's took place in the laboratory, as they should—and there were plenty of them. Unlike, say, preamplifiers, loudspeakers haven't evolved yet to the point where they can hide their faults from measuring instruments. When a speaker doesn't sound right, it doesn't measure right—and since not a single one sounds *quite* right, not one measures quite right. But, gratifyingly, the ones that sound better measure better. Needless to say, more than one or two kinds of measurement are necessary, and the results are seldom cut-and-dried, being generally the reflection of deliberate design trade-offs and therefore subject to interpretation. Even so, we're convinced that if every speaker manufacturer routinely performed our simple tests (as we know for a fact nine out of ten don't), speaker design would be a far less haphazard affair today and the art would advance more surefootedly.

We must emphasize, however, that our main concern in the laboratory was to make sure we hadn't made a mistake in our listening evaluations—that what we're reporting here is based on some sort of reality rather than on irrelevant artifacts of our component chain or on the aural idiosyncrasies of our staff members. For that reason, the overall thrust of our measurements was qualitative rather than quantitative; in the majority of cases we went only as far as it seemed necessary in order to authenticate our conclusions and recommendations—not to determine by how many dB or Hz or msec a manufacturer ought to change his design parameters to end up with a better speaker. (We try to give our subscribers useful advice and reliable explanations, but for $28 a year we're not going to give any company a product research and development program.)

After measuring, we listened again, this time of course "corrupted" in our subjectivity by our newly gained knowledge. And then, in some cases, we went back to the laboratory once more, just to double-check certain observations and correlations. The remarkable thing about this process is that it's synergistic; one becomes a more discerning listener after having objectively verified in the laboratory some purely subjective impressions, and one measures with greater astuteness and more concrete results after having aurally zeroed in on certain sonic peculiarities. Pretty soon we began to wonder how anyone could possibly form a serious opinion about a speaker without going through these steps—forgetting that there was a time when we didn't quite do it this way ourselves. (Who says a critic must be born perfect and never undergo any development?)

**The lab measurements.**

We measured each speaker in both the frequency domain and the time domain. We're convinced, without any reservation, that one without the other yields an incomplete picture and can't be satisfactorily correlated with the overall subjective impression made by a speaker. We do lean, however, toward the time domain as the more important of the two—otherwise why should a live singer or a live piano heard through a half-open window all the way around the corner still sound unmistakably live instead of canned? The frequency response is certainly shot to hell that way; only the time response retains its integrity, since all components of the sonic information are still traversing the air path at the same speed. (When you get right down to it, there's no such thing as "frequency"—identical sine-wave cycles per unit time—in the real world: it's an artifact of the human mind. But there *is* time all right, and nothing remains the same from instant to instant.)

Enough of this philosophizing. Here's what we did:

We measured the "nearfield" frequency response of each driver in each speaker. This corresponds very closely to anechoic measurement, as already explained in the subwoofer article in our second issue. We also took the overall frequency response curve of each complete system and, in some cases, of pairs or subgroups of drivers within the system. Both linear and log sweeps were used in these measurements, and both axial and power response were investigated.

Whenever it appeared that single-frequency distortion would be relevant to our evaluation, we checked the output of individual drivers for THD at the frequencies in question.

We ran quite extensive tone-burst tests on each system, at many different frequencies, with both sine-wave and square-wave bursts.

Perhaps most significantly, we tested each system with single pulses (call them blips or ticks, if you like) ranging in width from approximately 1 msec to 0.1 msec, spaced so far apart that room reflections didn't enter into the picture. Both the oscilloscopic trace and the audible quality of such pulses clearly reveal time-response information not easily obtainable any other way. The differences from speaker to speaker are absolutely startling on this test: some will reproduce a very adequate copy of the input pulse, with no garbage in the silent interval that precedes the next pulse; others will generate an output totally unrelated to the input.

In addition to these basic tests, we also listened to (and occasionally spectrum-analyzed) white noise and pink noise through the speakers, checked for structural faults, investigated impedance characteristics when we suspected significant peculiarities (but only then), and performed other little peripheral testing chores. We must repeat: our emphasis was on insight rather than on the accumulation of raw data.

The laboratory instruments we used were the Bruel & Kjaer 4133 calibrated microphone, the Hewlett-Packard 3580A spectrum analyzer, the Hewlett-Packard 1740A oscilloscope, the Wavetek 185 function generator, the Sound Technology 1700B distortion analyzer (with built-in oscillator), the General Radio 1382 random-noise generator, and the Hewlett-Packard 3435A digital multimeter.

**What about our usual sweeping generalizations on design criteria?**

Designing an electroacoustic transducer isn't at all like designing a preamp or a tuner. No one has ever come even close to theoretical perfection in a full-range speaker system, so it would be wrong to assert categorically that successful design must proceed in this or that particular direction. Somebody would promptly go the opposite way and probably do just as well. We therefore refuse to take sides on electrostatic vs. electrodynamic drivers, crossoverless vs. multiple-crossover systems, sealed vs. vented boxes, or any other design approach that can be either 95% right or 95% wrong, depending on how it's executed. We *will* take sides when it comes to the basic laws of nature (say, a claim of deep, loud bass out of a ¼-cubic-foot minisystem) or obviously wrongheaded engineering (like an incorrectly tuned vented enclosure, for example).

We rashly promised in our second issue

that we'd go more deeply into crossover design considerations in this survey, but the subject happens to be a bottomless pit. No sooner do we come to the conclusion that high-order networks with their very steep slopes can't possibly be any good on account of their nightmarish time-delay characteristics, one of our consultants comes in with mathematical proof that with certain kinds of time compensation they can be made to pass a perfect square wave. Of course, he adds, no commercial design is likely to take that route, because of its ridiculous complexity and high cost! And so it goes.

Since each speaker design turns out to be a law unto itself, our only choice is to make our generalizations, such as they are, under the individual product headings. So—we'll let the show begin.

# Acoustat X

*Acoustat Corporation, 4020 North 29th Avenue, Hollywood, FL 33020. Acoustat X full-range direct-drive electrostatic speaker system, $1995 the pair (with built-in power amplifiers). Five-year warranty, excluding tubes; customer pays all freight. Tested #491 and #492, on loan from manufacturer.*

Once again, the first piece of equipment in our alphabetical listing poses a special problem. We approach this review of the Acoustat X with great trepidation. The subject seems to be booby-trapped for the reviewer.

Why? Because, long after the completion of our tests, it remains unclear to us whether our samples were absolutely typical. Objectively, we must conclude that they were indeed typical and that our findings are applicable across the board. Yet, there's a persistent mystique to these speakers out there in audiofreak country, suggesting that maybe (just *maybe*) occasional super samples appear in circulation. The manufacturer professes to have no knowledge of such variability. Let's marshal the evidence:

Last winter, we heard the Acoustat X at a nearby store and were quite favorably impressed, although we did notice some peculiarities. We briefly reported so in our very first issue. In retrospect it seems that this was a better sound than we were subsequently able to obtain from our test samples, but then retrospect is a notoriously poor witness. We have since quizzed every audio enthusiast known to us who has had any experience with the Acoustat X, and their reports are wildly contradictory. A number of aficionados, including professionals with absolutely impeccable credentials, are completely sold on the Acoustat. State of The Art, they say. Others, equally expert—and these, unfortunately, happen to be the ones whose ears we have always trusted and who listen the way we do—have a much lower opinion of the speaker. The clincher came at the Summer CES in Chicago. The pair demonstrated in the Acoustat exhibit (presumably under the watchful eye of the manufacturer) sounded exactly—but *exactly*—like our test samples. As a matter of fact, a well-known and highly respected European record producer, whom we had met earlier, drew us aside as we were leaving the room and whispered: "Is *this* the famous Acoustat *electrostatic*? Does it sound right to *you*?" We rest our case.

To get down to business, the Acoustat X is a four-foot high dipole radiator consisting of three long, narrow, side-by-side panels, slightly angled with respect to one another, the middle one a bit narrower than the outside ones. The panels are full-range electrostatic transducers, driven directly off the plates of the 6HB5 output tubes of a matching hybrid power amplifier, which screws into the base of the speaker frame (or call it cabinet). Left and right channels are completely identical. On paper, the whole thing is a purist's dream: all the advantages of electrostatic (i.e., force-over-area) drive; no crossovers, hence no phase and time-delay problems from that source; no transformers in the signal path, as in most other electrostatics; no signal-processing stages between the preamp output and the ear that could be further eliminated. And yet . . .

All right, first the good news. The Acoustat X *does* have the unmistakable midrange integrity of a crossoverless electrostatic design. As long as the program material stays in a relatively narrow frequency and dynamic range (like, say, a singer with acoustic guitar), the sound is quite natural and likable. That alone may possibly account for all the favorable reactions. (Including ours in the dealer's showroom.) But feed the speaker some wide-range symphonic material, with banging timpani, clashing cymbals and soaring strings, or some heavily produced rock, and the naturalness is gone—even at quite reasonable volume levels. The speaker begins to sound strained and steely on top, bumpy and muffled below. The free-breathing ease and openness that characterize the reproduction of less complex material are replaced by a disturbingly canned quality. Vigorously played strings sound "electronic" even when they aren't massed, as in a trio or quartet. So does the piano.

We believe we know the reason for this behavior. Our tone-burst tests revealed substantial ringing at a number of frequencies, especially around 1 kHz but to some degree also around 4 kHz and several other spots. The energy peaks of complex, dynamic program material seem to excite these problem areas to the point where the ringing becomes distinctly audible and the sound quality suffers. The most plausible cause of the trouble would appear to be standing waves in the large panels, especially since the ringing is definitely lower in amplitude near the clamped edges.

Interestingly enough, single pulses are also very difficult to reproduce accurately through the Acoustat X. A crossoverless, force-over-area system could be expected to have all the advantages when it comes to this time-smear test (cf. the Beveridge below), but the Acoustat panels fail to render a good likeness of the pulse shape and are altogether unable to reproduce certain pulse widths (such as, for example, 0.35 msec). Whether this is symptomatic of the same problem as the ringing or unrelated to it we aren't ready to assert; it was just about at this point that we ran out of R and D time for the Acoustat Corporation. In any event, the Acoustat X lacks the ultimate definition, focus, and delicacy of inner detail that characterize a speaker with outstanding pulse response.

In the frequency domain the speaker is quite good, though not spectacular; overall power response is reasonably flat; there are, however, some very nasty lobes in the polar response. As a matter of fact, vertical beaming is so severe that it's hopeless to listen to the speakers standing up: your ears must be approximately level with the midpoint of the panels, otherwise you don't hear the highs. We could live with that, though, if the speaker had no worse faults.

One of these worse faults is in the deep bass: a great big bump in the response at 35 Hz that creates utter chaos whenever it's excited. The speaker actually *sizzles* when the signal generator sweeps through this bump, and organ pedal notes or bass-drum whacks anywhere in that vicinity overload the speaker to the point of offensive flatulence. It's even possible that the higher harmonics of this powerful resonance are responsible for that bumpy, muffled quality we discerned in the upper bass and lower midrange on complex program material. In addition, both the speaker frame and the amplifier chassis buzz at various frequencies. It's rather a mess.

Much has been made of the sensitivity of the Acoustat X to room placement, of the touchiness of its high-frequency balance control and even of its amplifier gain control, of the pros and cons of the built-in equalizer network that counteracts the boundary effect of the rear wall (it can be disabled by padding with a 470K-ohm resistor), of the desirability of raising the speakers off the floor, and so forth and so on. All we can tell you is: that's not where it's at. All these things make a difference, but not the Big Difference. We experimented with them all, staying in close touch with the manufacturer (who even sent us "new and improved" IC's for the front end of the amplifier—they were indeed better). We ended up using the speakers exactly as the instructions specify: 2 to 3 feet out from a fairly live wall, looking into the longer dimension of the room. That was by far the best arrangement, but they still didn't sound like reference speakers for the purist.

It must be remembered, of course, that the Acoustat X isn't really expensive—not when you consider that the power amp is included. That puts it almost in the same price category as the Dahlquist DQ-10; certainly not a whole lot higher. On the other hand, there's always the possibility that it's the amplifier that's causing some of the problems we observed. We doubt it, though. Since neither the speaker nor the amplifier alone is compatible with other equipment, the question is academic, and we didn't investigate it further.

Oh yes. We forgot. Both amplifiers eventually conked out in the course of our testing. One in the middle of some loud music, the other in the middle of a tone burst. Neither responded to elementary first aid. But by then our conclusions were firm. And will remain so unless we're presented with totally new evidence to the contrary.

# Beveridge Model 2SW

*Harold Beveridge Inc., 422 North Milpas Street, Santa Barbara, CA 93103. Beveridge Cylindrical Sound System, Model 2SW, $5200 the pair (with subwoofers, plug-in power amplifiers and control module). Virtually unlimited guarantee ("you must be absolutely satisfied"). Tested #233 and #234, on loan from manufacturer.*

Let's say it right at the beginning. This speaker/amplifier system, despite its undeniable limitations, comes closer to our ideal of sonic accuracy than any other we've ever heard, with the possible exception of Mark Levinson's stratospherically priced HQD System (which isn't really a single product but an *ad hoc* blend of four different brands and which we haven't tested yet under our own roof). After that statement has duly sunk in, we must add that some audio perfectionists will never settle for a system that can't play louder and has no more headroom than the Beveridge. Not at this price. To us, however, its specific virtues are irresistible.

The SPL/headroom/dynamic-range problem was of course the very reason why Beveridge didn't stick with the original Model 2, which was "purer" conceptually: a monolithic, crossoverless, full-range electrostatic system, with matching direct-drive high-voltage amplifier (a la Acoustat X but vastly more sophisticated, as we shall see). The Model 2SW, by incorporating a 70-Hz crossover (6 dB per octave and built right into the power amp) and letting a conventional electrodynamic subwoofer handle the lower bass, can play considerably louder, since the frequency division permits the electrostatic main unit to be optimized for the narrower range. Even so, forget about any SPL above 100 dB or thereabouts in a fair-sized room. The Beveridge sounds so good when it doesn't have to strain that it's a crime to strain it. (Besides, the fuse will blow.)

Why does it sound so good? There are two main reasons, in our opinion. One is that it's probably the only speaker that's a *true* line source and lays a sound field into the room accordingly. (The Infinity "Quantum Line Source," for example, isn't and doesn't.) The other is its incredibly good impulse response, indicating extremely low time-delay distortion, which audibly outweighs whatever minor deficiencies the speaker may have in the frequency domain.

A quick description of the design: A 6-by-1-foot electrostatic transducer is front-loaded by a 6-foot high "acoustic lens" terminating in a relatively narrow aperture. The rear radiation is blocked. The lens, which is in effect a system of vertical guiding walls, bends the planar wave produced by the electrostatic panel into a 6-foot high 180° cylindrical wave front, invariant with frequency, in accordance with the geometry of a line source. The transducer, the lens, and the baffle structure for the back wave are all housed in a huge, upright, coffin-like structure that plugs directly (through contact pins in its bottom) into a metal base that houses the amplifier. The system is best set up with the apertures of the left and right units facing each other across the room. The two subwoofers (large but manageable) plug into their respective amplifiers via ordinary speaker wire and banana plugs. A 30-foot shielded cable emerges from each amplifier to be plugged into your preamp. A separate control module is also supplied with the system, with provisions to trim the frequency response within a few dB ("spectrum slope" and "bass environment") and to vary the stereo spread ("lateral control"). We found that this control unit introduces an ever-so-slight (call it infinitesimal) degree of veiling, so we unplugged it, especially since our listening-room characteristics happened to suit the naked speakers very well. On the test bench, the control unit gave no evidence of any type of distortion; indeed, it's very well designed.

The sound of the Beveridge Model 2SW is such that, at a reasonable SPL and with superior program material, most listeners go "yechh" when any other speaker is switched in for comparison. Even the good ones in our survey. The Beveridge is far superior in transparency, midrange and (especially) top-end openness, freedom from nasality and vowel-type colorations, depth perspective, attack and release, and overall delineation of inner detail. Furthermore, that cylindrical wave front really does what it's supposed to; uniform distribution of the direct-path "first-arrival" sound results in absolutely unvarying balance and perspective anywhere in the room—near or far, left or right, sitting or standing, or even right between the speakers. Since the SPL from a line source varies inversely with the distance, not the *square* of the distance as in the case of conventional speakers, the uniformity of the sound field is even more startling. Stereo imaging is also excellent and quite independent of listener position, but be prepared for differences in comparison with good point-source speakers, such as the Rogers. The fact is that the exact complementary playback geometry for a particular microphone setup is always conjectural and therefore highly arguable, so that the characteristic wall-to-wall stereo image created by the Beveridge is as valid as any and at least never smeared by time delay.

What all this adds up to is a strikingly original design concept executed with a high degree of competence. Not since the Lincoln Walsh speaker has there been anything comparable in sheer creative thinking—and the actual implementation of the Beveridge idea is of course far superior (cf. the Ohm F below). It's interesting to note that Harold Beveridge, who is responsible for both the theory and the practice, never had anything to do with the audio industry, having spent his entire engineering career in high-level national defense jobs. It seems that hanging out with Bill Johnson or Joe Grado or Dick Sequerra isn't the only way to learn about the frontiers of audio.

We must now point out what's wrong with the speaker. For one thing, the woofer is only pretty good instead of great, which is what the Beveridge electrostatic would deserve. The 12-inch woofer in its sealed enclosure appears to have a system resonance of 36 Hz, since at 40 Hz its unequalized amplitude response is up 5 dB, indicating a Q of 1.7. The response drops back to the 0 dB level at 30 Hz. This is not the way to extend response down to 30 Hz, in our opinion; in fact, it doesn't *sound* like 30 Hz response but rather like bumped-up 40 Hz response, which is of course what it really is. For example, the open E string of the double bass, vigorously plucked, makes the woofer go "boom" instead of "thunk"—and that's almost exactly 40 Hz. It wouldn't happen with a Q of 0.707. Harmonic distortion is again pretty low but not fantastic; the original Acoustic Research woofer of 1952 was better in that respect. (Ed Villchur, where are you now that we really need you?)

There's also quite a nasty resonance at 76 Hz in the large (electrostatic) enclosure. It's a fairly broad peak of about 8 dB amplitude and you can hear it all the time if you listen carefully.

*Editor's Note:* **At press time we're told that Harold Beveridge has identified the problem as due to insufficient damping material behind the electrostatic panel, has cured it completely (to 0 dB), and will make the modification in our review samples at the earliest opportunity. Not in time, however, for this already late issue.**

The frequency response of the complete system is otherwise very smooth to 12.5 kHz, displays some slight perturbations in the 13-14 kHz region (probably due to standing waves between the lens elements), peaks very gently at 16 kHz and then dies quite fast. The farfield integrated response, however, smooths out these minor, and to our ears inaudible, irregularities, so that the picture is one of a very gradual downward slope from 12.5 to 20 kHz, without significant peaks. Overall, we have no quarrel with the frequency response of the Beveridge from, say, 100 Hz on up.

In any event, it's the time response of the speaker that puts it in a class by itself. It reproduces single pulses of just about any reasonable width with startling accuracy. And you don't have to hunt for a "sweet spot" to read a good pulse, as you do with even the best conventional speakers; just point the measuring microphone in the general direction of the speaker—high, low, sideways, over your shoulder while talking on the telephone—and the pulse will be a good replica of the generator output. Again we must say—that cylindrical wave front sure does its job.

One more thing. We haven't heard the original, simon-pure Beveridge Model 2, so we can't comment on the similarities and differences. According to Harold Beveridge, however, the overall gain in headroom is almost 10 dB with the Model 2SW, which would totally disqualify its predecessor in our estimation, since even the Model 2SW is just adequate in that respect. In fact the I-think-I-just-heard-it-clip crowd will find it rather frustrating, as we've already said.

To us, the speaker remains a joy. It's the only one in this survey that can occasionally make us forget that we're listening to reproduced music. Now and then, it almost sounds live. And that makes it our reference speaker, until somebody sends us something better.

# Braun 'Output C'

*Adcom, 114 East 32nd Street, New York, NY 10016. Braun 'Output C' miniature speaker system, $224 the pair. Tested samples on loan from distributor.*

Imagine a speaker system no bigger than one of the heftier twin-lens reflex cameras and in black metal, too. Imagine it with sound that's quite respectable for any speaker, regardless of size: just a notch or two short of top-notch and minus the deep bass. You now have a fairly good picture of the Braun Output C. Amazing and lots of fun.

We wouldn't hesitate to recommend Braun's small wonder to anyone who is really short of space and doesn't even have room for anything like the little Rogers LS3/5A (which is far superior but about three times as big and twice as expensive). The Output C is also the perfect gift "for the audiophile who has everything." A great toy.

Don't let anyone tell you, though, that these diminutive speakers plus a commode-type subwoofer with summed channels constitute an "invisible" system approaching SOTA quality, as has been suggested in some quarters. No way. The Output C has some definite colorations even in the range where its tiny woofer and dome tweeter are reasonably flat. These colorations, however, aren't particularly irritating musically; the overall sound of a pair of Output C's used as a full-range stereo system is open, smooth and listenable beyond all expectations. Even the bass is subjectively acceptable, although it's mostly fake of course, achieved with bumped-up response at the system resonance of 200 Hz rather than with genuine low-frequency output.

The most noticeable coloration is due to the tweeter, which rings rather badly and doesn't allow the system to reproduce pulses accurately, especially as the pulses get narrower. The result is just the slightest zippiness and a lack of the utter transparency you can expect with superior impulse response (cf. the Rogers).

Even so, we have nothing but admiration for the Braun Output C. The very fact of its existence is a sign of progress in audio engineering.

# Braun L200

*Adcom, 114 East 32nd Street, New York, NY 10016. Braun L200 small bookshelf speaker system, $260 the pair. Tested samples on loan from distributor.*

These neat little Braun speakers are somewhat anticlimactic after the Output C's; they're about twice as large (i.e., still exceedingly small) but not much better sonically, so the amazement factor is considerably smaller.

The differences: black plastic rather than metal enclosure, larger woofer of course, same type of dome tweeter but not quite the same diaphragm material. The system resonance is at 175 Hz instead of 200, bumped up the same way (the Q is approximately 1.5). The tweeter still rings, pulse reproduction still deteriorates as one goes from wider pulses to narrower ones, but there's some improvement in these areas. The overall sound is perhaps somewhat rounder and more authoritative, but not very different. The slight coloration is still there.

If it were possible to avoid comparing the L200 with its little brother, we'd be impressed; under the circumstances, it's the Output C that stands out as the more remarkable product.

# Canton LE 400

*Adcom, 114 East 32nd Street, New York, NY 10016. Canton LE 400 bookshelf speaker system, $350 the pair. Five-year warranty; manufacturer pays return freight. Tested #466872 and #466888, on loan from distributor.*

Through an alphabetical coincidence, the three German sealed-box minisystems in this survey follow one another without interruption. The Canton LE 400 is by far the largest: about half a cubic foot in volume. You'd expect it to have better bass than the two little Brauns and it does; the system resonance is at 110 Hz, where the response is up 6 dB, meaning that the Q is approximately 2, which is definitely on the loosey-goosey side. The corner frequency of the bass roll-off is 74 Hz, as near as we can tell. The manufacturer claims response down to 35 Hz, which of course is nonsense.

There's actually a three-way system shoehorned into that tight little box, with crossovers at 750 Hz and 2600 Hz. Frequency response is extremely flat from about 700 Hz on up. The star performer is the tweeter, which has astonishingly flat response on axis to well above 20 kHz and even at 45° off axis goes out past 15 kHz. That's superb amplitude response and dispersion in any league. It's also the most likely reason why the speaker system sounds impressively open and "present" on first hearing.

We say first hearing because, after a while, the LE 400 begins to sound overbright, "electronic" and fatiguing. The culprit is again ringing, easily identified by tone bursts, especially at 2 kHz and 4 kHz. Pulse reproduction is also surprisingly poor.

If you have a chance, though, listen to this speaker. It's the least expensive we've ever heard that, if only for a minute or two, has the very special sound of the top systems with fast tweeters. Then it's all over, but we're willing to bet that the sale is often made by then. Cute little rascal.

# Cizek Model #1

*Cizek Audio Systems, Inc., 15 Stevens Street, Andover, MA 01810. Model #1 acoustic-suspension loudspeaker, $396 the pair. Five-year warranty; manufacturer pays all freight. Tested #2054 and #2065, on loan from manufacturer.*

This is one of the medium-sized units in our survey: a solidly built acoustic-suspension system of the largest bookshelf type (almost too large for a bookshelf), with 10-inch woofer and 1-inch dome tweeter. Despite its conventional format, it comes with a special audiophile reputation, since the company is technically oriented and the speaker is more sophisticated in engineering as well as in audible performance than such speakers generally are.

Its best feature is its bass response; the system resonance is at 39 Hz (the specs say 38 Hz, which is accurate enough for us), and the damping is absolutely correct. In fact, at the flick of a switch on the front panel, you can have a Q of 0.6 for dead-flat bass that drops to -4 dB at resonance, or you can choose a Q of 1 (just a little looser but still well-controlled) for 0 dB response at resonance and the slightest ripple just above. These people know their P's and Q's; they also know better than to try to squeeze more out of a given woofer or box size than Mother Nature permits. As a result their bass is close to state-of-the-art within the physical limitations of the system—solid, well-defined, musical. It's so easy when you do your homework.

The most highly touted part of the speaker is the crossover network, which is made to see a resistive rather than a reactive load by means of additional impedance compensating networks across each driver. Cizek claims that there's no other way to achieve absolutely flat response in the crossover region, which is a sensitive part of the spectrum in this case, the nominal crossover frequency being 1.5 kHz. Another sophisticated touch is the combination of two high-frequency controls on the front panel (right next to the Q control), one for contour, the other for overall level. Their net effect is rather subtle and, in our opinion, something of a red herring in view of more serious matters in the high-frequency area that require our attention. The least of these is a 5 dB peak at 16 kHz with a precipitous drop thereafter, which is what we blamed initially for imparting a somewhat irritating zing to the otherwise smooth and classy sound of the Cizek. That wasn't it, though. It turned out to be much heavier stuff.

Exploring the speaker with tone bursts, we came upon an astonishing amount of ringing at 3 kHz. It seems that the tweeter reproduces the burst quite accurately, after which the much slower woofer, still quite active at that frequency, chimes in with another complete burst of almost the same amplitude. Two bursts for the price of one: you feed in, say, a four-cycle burst and out comes an eight-cycle burst with just the tiniest cleft in the middle. Needless to say, at 3 kHz, which is right where the ear tolerates absolutely no nonsense, this spurious doubling of energy sounds like excessive brightness or glare. The ringing continues higher and lower, too, but 3 kHz is its headquarters.

Our pulse tests confirmed the lack of synchronism between woofer and tweeter to be the speaker's chief weakness. The tweeter is much too fast for the woofer—or, if you prefer, the woofer is too slow for the tweeter. First the tweeter blips, then the woofer; you can actually hear the double blip, unless you put the woofer so much closer to your ear that the speaker, laid on its side, faces inward instead of out toward the listening area. The microphone, too, reads the best pulse from this extreme off-axis position. A time-compensating network could probably do wonders for the Cizek.

Where does all this leave us as regards an overall evaluation? We still feel that the Cizek is one of the very best box speakers anywhere near this price. Not because it's without faults but rather because none of the others we're aware of have acceptable full-range performance. The Rogers, for example, sounds incomparably better but can't handle any power and lacks almost two octaves of the Cizek's bass range. It also costs more, at least in the U.S.A. The DCM Time Window is also significantly better but can hardly be called a box speaker and is priced very much higher. And so on.

Maybe we shouldn't have reviewed the Cizek in such fast company. Still, we feel that the bass alone is worth the price of admission and that the basic design is not only intelligent but also capable of evolutionary improvement.

# Cizek Model #2

*Cizek Audio Systems, Inc., 15 Stevens Street, Andover, MA 01810. Model #2 acoustic-suspension loudspeaker, $268 the pair. Five-year warranty; manufacturer pays all freight. Tested #1130 and #1139, on loan from manufacturer.*

Speaking of evolution, Cizek has already come up with a good example: their Model #2 is a considerably smaller and less expensive version of the Model #1 while still offering the same basic sound quality.

The main difference is the substitution of an 8-inch woofer for the 10-incher; in combination with the smaller enclosure volume this still keeps the system resonance at the same frequency, namely 39 Hz (or 38 Hz, according to Cizek). In fact, the Model #1 and Model #2 have virtually identical bass response profiles; the two-position Q adjustment is also retained on Model #2, except that the obtainable values are 0.57 and 0.9, certainly not a major difference. The tweeter and crossover network appear to be the same, but only a high-frequency level control is provided, without the contour control.

Both models sound extremely similar. On organ music and other complex program material that's rich in bass energy, the Model #2 poops out much more readily; it just can't handle power the same way and begins to distort. On the other hand, it has slightly less zing and glare at the higher frequencies—and there's a reason. The peculiar multiplication of tone-burst cycles occurs at around 2 kHz on Model #2, which is less irritating than 3 kHz ringing. Furthermore, the amplitude of ringing is quite a bit smaller. The 16 kHz peak is still there, though, and so is the differentiation of pulses everywhere except way off axis toward the woofer side. It's basically the same speaker.

And that makes it pretty good value in our book. Purists, on the other hand, need not apply.

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***Editor's Note:*** **The latest Dayton Wright full-range electrostatic speaker, announced for this survey on the back page of our last issue, should have followed here in alphabetical sequence. Unfortunately, it wasn't sent to us in time for testing, despite a firm commitment from the company. We now have another firm commitment from them to make the speaker available in time for Part II of the survey in our next issue. If they don't come through, we'll try to obtain a pair anyway, one way or another. Meanwhile, on the basis of admittedly superficial exposure to the Dayton Wright in four different places at various times, we're inclined to doubt that it would challenge the Beveridge as our top choice. But we're perfectly ready and willing to change our mind should our tests turn out to be contraindicative.**

* * *

# DCM 'Time Window'

*DCM Corporation, 725 S. Division, Ann Arbor, MI 48104. 'Time Window' floor-standing loudspeaker, $660 the pair. Tested #906 and #932, on loan from manufacturer.*

Here's the one that will recoup you for about 20 years' subscription to **The Audio Critic**: a speaker system that has no equal even at twice the price (at least none known to us). In this Part I of our survey, we place only the following speakers ahead of the DCM Time Window: the Beveridge Model 2SW, the Rogers LS3/5A (but only from, say, 120 Hz on up at moderate levels—or else with expensive subwoofers), and the Koss Model One/A (but only every other day because in many ways we prefer the Time Window). That's all. The Dahlquist DQ-10, for example, which was our reference speaker before we started the survey, ranks below the Time Window in our current hierarchy.

There's some consistency to the above: every speaker that, to our ears, equals or surpasses the Time Window has outstanding response in the time domain. The Beveridge reproduces pulses most accurately; the Rogers is next; the Time Window and the Koss are close behind the Rogers and about equally good, but with quite different deviations from perfection. How about that? You'd swear that we measured impulse response first and then picked our favorites, but you have our word of honor that our preferences were firmly established prior to any laboratory tests, strictly on the basis of listening. We have a feeling that, in the not too distant future, speaker designers will be investigating impulse response as routinely as frequency response.

The DCM people are among the most fervent advocates of pulse testing; the very name of their speaker, Time Window, shows where their head is at. Their literature makes the most convincing case we've seen for the overwhelming importance of the time domain; if you discount its quite mild and inoffensive commercialism, you can learn more from it than from just about any other popular explanation of speaker design. It shows comparative pulse tests on fifteen speakers, including the Time Window, and even if they manage to make their own product look a little better than *our* test results indicate, the brochure is an eye-opener.

The physical configuration of the Time Window is quite unconventional. Imagine a waist-high cylinder standing on the floor, except that only the back of it is cylindrical; the part facing you, the listener, is a triangular prism, edge foremost. On the two angled faces of the prism are the drivers, symmetrically placed: two 6-inch Philips woofers and two Philips dome tweeters below them (that's right, below the woofers, not above them). Near the floor are two ducted ports, also symmetrically placed. The cabinet is surprisingly light; you can carry it from one end of the room to the other without requiring abdominal surgery afterwards, yet it's quite rigid. Weird but effective design.

The proof of design is of course in the listening, and what a nice-sounding speaker it is. The first thing that strikes you is truly convincing spaciousness and depth; when a singing chorus walks in from offstage in an opera recording, you can almost tick off the yardage as they approach. Only the Beveridge is in the same league in this respect; the Time Window beats the Koss and all the others. Transparency and delineation of inner detail are of a high order, but not quite up there with the Beveridge or even the Rogers; the Koss, too, sounds more open and delicately etched in the upper octaves but not throughout its range. Left/right imaging isn't quite as spectacular as with, say, the Pryamid 'Metronome', but we find it musically satisfactory. In fact, the entire speaker sounds unfailingly musical at all times and remains listenable regardless of your length of exposure to it.

That doesn't mean, however, a complete absence of colorations or other audible anomalies. Far from it. The bass is a little funny for one thing; more about that in a moment. The top end could be a little more extended and silky-smooth; that last touch of HF quality seems to be unattainable with the Philips dome, even though it's better than most and handles power very well. There's some raggedness in frequency response throughout the audio range, which may be the reason why the speaker is particularly sensitive to room placement; you've got to make sure that the room reflections zig where the speaker zags. It isn't anything serious (minor frequency-response problems seldom are), except in the bass and the lower midrange, where things get a bit thick and muddled from time to time. Carefully tuning the distance of the speakers from the back wall as well as the side walls will clear up the problem; raising the speakers a foot or so off the floor also helps. When everything is trimmed in, the lows are quite clean and tight, but don't expect stupendous pressure bass on organ pedals, bass drum or bull fiddle. If we were pressed to designate a nominal corner frequency for the Time Window's rather bumpy *composite* bass roll-off, we'd place it at 50 Hz. Maybe even a little higher. After all, it's not a large box.

We say "composite" because the bass response of a vented box is the sum of the outputs produced by the woofer and the vent. Ideally, these should be complementary, arithmetically adding up to flat response. In the case of the DCM Time Window they aren't. Each woofer exhibits the classic vented-box response you'd expect, but the vents by themselves have *perfectly flat* output down to about 18 Hz instead of filling in for the woofers with a humped response where the woofers drop out. Somebody obviously thought that this was a very good thing, but of course it doesn't add up to flat bass response out of the total system. The DCM design philosophy, as we understand it, is that in order to generate flat response within four walls, a speaker should never be designed for flat response anechoically (i.e., with nearfield measurement). We disagree with this philosophy quite vigorously and feel that it may have a great deal to do with the temperamental behavior of the Time Window as regards room placement. We don't want to make too much of the whole thing, on the other hand, since the speaker is still a lot better with this flaw than others are without it.

The reason why it's better—why it *sounds* better—is that (a) it reproduces pulses very accurately, with only minor glitches, and (b) it exhibits little or no ringing on tone bursts. In other words, it doesn't smear the signal. (One of those minor glitches worth noting is a little negative blip out of the tweeter just before the leading edge of a positive-going 0.1 msec impulse. We mention it only because it's very neatly fudged in the DCM literature; visible but cleverly buried. You can also tell from their pictures that the various electrostatics don't have this problem, thus confirming each of our own findings. Naughty, naughty.)

To sum up, the DCM Time Window is one of our happiest discoveries since we've started testing audio equipment, and we recommend it wholeheartedly to any music lover who isn't planning to spend thousands of dollars on his speaker/amplifier setup. It makes sense, however, to get a very good amplifier to drive the Time Window; not necessarily a very high-powered one, since the speaker is quite efficient, but a really clean one because superior time-domain resolution will show up the difference.

Our congratulations to Steve Eberbach, the young engineering partner of DCM, for what may very well turn out to be a classic.

# Fundamental Research

*Fundamental Research, 1304 Success Street, Pittsburgh, PA 15212. 'The Low Frequencies' subwoofer, $450. Tested samples on loan from manufacturer.*

This is the best subwoofer we've tested so far, which by itself isn't a particularly great distinction, since the Janis W-1 and W-2 are badly flawed designs and the Dahlquist DQ-1W doesn't go low enough. The Fundamental Research is a good unit, however, in any company; that's why we didn't hesitate to include it in this survey. And it happens to mate very nicely with the Rogers LS3/5A (more about that in the review of the latter below).

The Fundamental Research is a waist-high floor-standing subwoofer incorporating a 12-inch driver in a sealed box with an approximate internal volume of 3 to 3½ feet. It's perfectly classic and straightforward in design except for a weird little slotted board that's nailed right on the front baffle across the face of the driver. This doesn't provide any low-frequency loading but is there ostensibly to dissipate the higher frequencies—a very crude sort of termination. It's responsible for the one serious flaw in the response of the woofer: a peak at around 450 Hz that rises 7 dB above the flat portion of the curve, followed by considerable raggedness (huge suckout at ca. 1 kHz, double hump back up to the 0 dB line at 1.4 and 2 kHz, ragged roll-off thereafter). Ideally, a woofer should roll off smoothly above a given frequency (450 Hz would be perfectly all right), so that the crossover network can do its thing without running into unpleasant surprises. (Remember the 14 dB peak of the Janis W-1 at 460 Hz?) We must admit, however, that we could detect no audible anomalies above the crossover point when biamping the Fundamental Research either through the Rappaport PBC-1 passive crossover or the Dahlquist DQ-LP1 active/passive crossover, with the nominal crossover frequency in the 100 to 120 Hz region. It appears that the goof has to be of Janis proportions before it can be heard.

That said, we give this subwoofer a high mark for its bass response profile. Down to the roll-off point it's dead flat; the -3 dB frequency is 38 Hz; from there on the response drops to -5 dB at 30 Hz, -8 dB at 20 Hz, and -14 dB at 10 Hz. This is a slightly overdamped profile (the Q appears to be a little lower than 0.707), so that there's probably room for further improvement (i.e., a lower corner frequency without the penalty of overhang) with some fine-tuning of the parameters. If you've read our subwoofer article in the second issue, and if we now tell you that the Fundamental Research handles power quite well, you need no further information to know exactly what it sounds like. The bass is deep, solid, well-controlled—it couldn't be anything else. Mother Nature practices no deceptions. (Or, as one of our consultants is fond of saying, the spectrum analyzer is too dumb to lie.)

Overall, we prefer the Fundamental Research to the Dahlquist DQ-1W, even though it's much less efficient and therefore requires a bigger amplifier to drive it. But it holds up much better in the bottom octave and at the same time it's just as tightly controlled. Not quite as well built, though; it has a few cabinet buzzes, seldom excited even by loud music, that we'd rather do without. A little more wood or a little more glue seems to be called for.

We've left the best for the last. This subwoofer was designed purely *by ear* by a dedicated young man named Mike Zelenak. The chances that it would come out this successfully were slim, to say the least, but there it is —you can't argue with a historical fact. The moral is that, if your ears (and taste) are as good as Mike's, it's still possible in audio today to fly by the seat of your pants. Otherwise, especially if you're into woofers, stick with good old Thiele and Small. We will. But then we also believe in keeping road maps in our glove compartment.

# Infinity QLS

*Infinity Systems, Inc., 7930 Deering Avenue, Canoga Park, CA 91304. Quantum Line Source speaker system, $2500 the pair. Five-year warranty; customer pays all freight. Tested #7001070 (left) and #7001071 (right), on loan from manufacturer.*

In our first issue, we published a rather negative preview of the Infinity Quantum Line Source, based on superficial exposure to it in a dealer's showroom. After thorough testing under our own roof, we have somewhat more respect for this speaker, but by no stretch of the imagination and with no amount of good will could we call it State Of The Art, Reference Standard, or any other such name dear to the hearts of audiophiles.

To put things into perspective, if our first exposure to the QLS, as inserted into our reference system, had been with a blindfold on, we would have said, "That's a fairly decent-sounding speaker system; what is it?" On being told that it's Infinity's current flagship, successor to the fabled Servo-Statik and $1250 per side, we would have said, "You must be kidding!" When properly set up and balanced out (which takes some doing), the QLS reproduces the full audible range of frequencies with a certain degree of authority and musical felicity. But it never reveals sufficient inner detail (the pleasure derived is more in the area of maple syrup than of crystals); makes the stereo image shift and drift depending on frequency; woofs up and occasionally even burps when excited by really fierce bass transients; in short, it just doesn't sound like one of the more accurate systems money can buy.

The reason for some of these shortcomings is obvious as soon as you remove the grille from the front panel of this 5½-foot high, 150-pound monument. Have you ever seen or heard of a speaker with 17 (count them) drivers that was an accurate reproducer? We haven't. It's impossible to synchronize that many separate sources of radiation to produce a single, unconfused image. Infinity seems to believe that by stringing out eight tweeters and six midrange domes in two *parallel* vertical lines they've created a Line Source (capital initials theirs). Anyone who has read our Beveridge review above has some idea of what a line source really is and does. Where's the cylindrical wave front that's invariant with frequency? Where's the time coherence? We found that the QLS reproduces pulses quite poorly, lacking even an acceptable "sweet spot," which is totally contradictory to the definition of a true line source.

What's more, the responsibility for balancing out those 17 drivers is left entirely to the user. There are three continuously variable level controls in the back (midbass, upper midrange, tweeters) with a rather wide latitude of adjustment; in addition, the tweeters can be brought in at three different crossover points by means of a plug-and-socket arrangement. Assuming conservatively that the continuously variable controls have only four audibly distinguishable positions each (say, all the way down, all the way up, and two in between) the total number of different-sounding response profiles possible is 192. The speaker designer is saying, in effect, "I don't know how the hell it should sound. You tell me, baby." There's a point where flexibility becomes anarchy. We tried our very best, but we can't guarantee that further knob-twiddling wouldn't have resulted in slightly better sound in our room.

Even with the help of a spectrum analyzer and a calibrated microphone, the overall frequency balance (i.e., farfield amplitude response) of the QLS is extremely difficult to determine on account of those 17 drivers strung out all over the place. All we were able to do was to analyze the tweeters, the midrange domes, the "midbass coupler" and the woofer separately.

We must admit that we were very impressed by Infinity's samarium-cobalt Electromagnetic Induction Tweeter (EMIT), a force-over-area transducer claimed to possess all the advantages of electrostatics without their drawbacks. We found no evidence to contradict that claim; the EMIT appears to be extremely smooth and flat out to 23 kHz, with wide horizontal dispersion. We haven't seen or heard a better tweeter, and that includes the rather similarly configured electrostatic tweeter in the Koss Model One/A. In fact, we got the most musical sound out of the QLS by selecting the lowest available tweeter crossover point and assigning the widest possible range of frequencies to the EMIT. That way the highs remained silky and sweet no matter what we fed into the speaker; furthermore, the EMIT reproduced pulses accurately as long as it was tested all by itself, with the adjacent midrange source blocked out. Now all that Infinity needs to do is develop an EMIT that goes all the way down to 20 Hz!

The midrange dome used in the QLS is, on the other hand, a real turkey; it has, among other things, a huge peak at 2.2 kHz, which is probably one reason why the QLS doesn'thave a more transparent upper midrange. As an interesting sidelight, this very same Peerless driver (Peerless is owned by the same parent company as Infinity) is available to all comers in the industry *without* the little pinholes in the dome that Infinity insists on having in the QLS version. To us the latter looked from the beginning like a cute little Helmholtz resonator—and sure enough, it resonates. We've heard it rumored that the Peerless people themselves don't quite understand why Infinity wants holes in the dome. Peerless, incidentally, also supplies the single 5-inch midbass (200 to 600 Hz) driver for the QLS, and it just so happens that this is the other slow member of the family (the wages of incest?), ringing like crazy at 200 to 250 Hz and contributing a retarded me-too response to 0.1 msec pulses, which are pretty far outside its assigned range. Talk about time smear . . .

Which brings us to the woofer—the same Watkins woofer we referred to in our preview as well as in our CES report. This isn't a unit to be dismissed quite as peremptorily as we originally did, even though the uncomplimentary remarks about it that we attributed to a distinguished American professor and electroacoustician are confirmed to have been uttered substantially as quoted. Another heavyweight academician, the polymath Dr. Richard H. Small of the University of Sydney, Australia, takes a somewhat kinder view of the woofer, as revealed in a letter to William H. Watkins, the inventor, a copy of which was made available to us by Infinity. More about that in a moment. Our own measurements indicate dead-flat response down to the lowest reaches of the audio range; the -3 dB point is 22 Hz! No other woofer known to us comes even close to this kind of penetration of the bottom octave, except the Janis W-1; and, as in the case of the Janis, we must point out that the audible performance doesn't live up to the promise of the response curve. The low bass is quite beautiful, detailed and controlled as long as the energy level remains moderate; however, when the woofer is called upon to move a lot of air (as in the case of one of our master tapes on which two double basses are plucked fortissimo), some rather rude eructations ensue, even with the best and most powerful amplifiers available. This doesn't surprise us, since a 12-inch woofer, Watkins or no, just doesn't have the piston area and linear excursion capability to find happiness in a 7 or 8-cubic-foot sealed box as a 22-Hz system—not at high SPL, anyway. Either the box should be smaller and the bass roll-off higher (cf. the Fundamental Research above), or the driver should be at least a 15-incher. As is, the system is really reaching when it digs into that bottom octave.

Even Dr. Small, of whose letter Infinity seems to be inordinately proud, points out that the Watkins woofer has no large-signal advantage over a conventional design. When it comes to moving air in a room, the cone alone must do all the work, and the limiting factors are cone size and linear throw. What Dr. Small likes about the Watkins invention (a two-voice-coil configuration that eliminates the conventional hump in electrical impedance at the resonant frequency) is that under small-signal analysis it appears to provide an increase in output of approximately 2 dB, all other things remaining equal—a theoretical free lunch that appeals to the academic mind. The penalty is that the two-voice-coil motor sucks current from the typical constant-voltage amplifier like there was no tomorrow, and the extra juice had better be available if you want to enjoy that 2 dB advantage. (Our Bryston 4B gave us no problem.) The amusing thing is that Dr. Small, a generous man who knows ten times more about speakers than Watkins, Infinity and **The Audio Critic** put together, peppers his mildly condescending praise of the design with quite a few sly professorial knuckle raps, which make his letter something less than a show-off piece encased in plastic for Dad's wallet. Certainly his exegesis of the woofer is at considerable variance with Infinity's simplistic technical brochure on the subject, which treats the low-frequency impedance peak of a speaker system as some kind of foul disease for which a cure has finally been found. We have personally heard Neville Thiele, Dr. Small's original mentor, call that point of view "rubbish." An impedance peak and an amplitude response peak are two very different things. So much for the Infinity/Watkins/Small issue.

A word about the biamping option on the back plate of the QLS. If you connect separate power amplifiers to the woofer and to the rest of the system, without using an electronic or passive crossover ahead of the amplifiers, you gain very little in performance. Each amplifier channel will still see the full frequency range of the signal at its input and output, except that part of the output signal will be burned up in a network instead of being converted into sound. For reduced IM distortion and increased headroom, you need the additional high-impedance crossover. Of course, the fact that the separate terminals are there is a convenience. In any event, biamped or not, the QLS needs a lot of watts to drive it, mainly on account of that power-hungry woofer.

To sum up, then, the Infinity QLS gives you a big, juicy, quite pleasant and listenable sound, smooth and extended on top but far from transparent or analytical overall, with time smear quite evident and no stable image. All pretty much confirmed by measurement. Among the speakers we'd rather listen to are not only costlier ones like the Beveridge or competitors like the Koss, but also considerably lower-priced ones like the Snell, Dahlquist, DCM Time Window and Rogers LS3/5A.

# Koss Model One/A

*Koss Corporation, 4129 North Port Washington Avenue, Milwaukee, WI 53212. Model One/A full-range electrostatic speaker, $2100 the pair. Tested early production samples, on loan from manufacturer.*

Until the arrival of our Beveridge Model 2SW, the revised Koss electrostatic was more or less our reference speaker in the course of this survey. We say more or less, as we were never completely sold on it; however, in comparison with all the other speakers (except possibly the Rogers, which is a very special case), it just sounded more open on top, clearer, more analytical, less smeared. In other words, like a good full-range electrostatic. At the same time it had a disturbingly closed-down, thick, gag-in-the-mouth quality somewhere between the bass and the lower midrange—we couldn't decide exactly where. We experimented with room placement, biamping, taking off the back, you name it. Our lab tests finally revealed what we now believe to be the cause, but it wasn't something that could be helped without major surgery. We'll come back to that in a moment.

To understand the Koss Model One/A, you must realize that it represents just the opposite approach to full-range electrostatic design from the Acoustat, the Beveridge or even the Dayton Wright. It's much closer to the Quad, except for its four-foot height. The audio range is divided into four passbands and specialized electrostatic elements are designed for each, to be crossed over at 6 dB per octave. There are four large bass panels, one midrange panel (about the same size), one narrow treble panel, and an even narrower tweeter panel. They're driven via transformers by any amplifier of your choice. The main difference between the original Model One and the One/A is that in the latter the treble and tweeter panels are placed next to the midrange panel (at the inward edge of each speaker, to form a mirror-imaged L/R pair), instead of partially covering up the midrange panel with disastrous effect on the sound. Either way, there's no serious attempt to time-align the panels; basically they're just fastened into the frame to fill out the available space. We suspect that this is one of the fundamental design limitations of the Koss, since the superior time-domain characteristics of each individual panel don't add up to coherent time response by the total system, except over a very narrow angle, way off toward the outside edge. The speakers would have to be angled inward to take advantage of this, not just a little but to an extent that wouldn't be tolerated in most homes. Too bad; the pulses are truly excellent from that angle but deteriorate badly as the head-on position is approached. We aren't suggesting that this isn't preferable to time smear that originates in the transducers themselves; the One/A has audibly better inner detail than nearly all other speakers, but the design concept isn't taken all the way to the limits of its potential.

In the frequency domain the speaker is also very impressive; it's possible to find a "sweet spot" where the response from 1 to 17 kHz is simply dead flat. The treble and tweeter panels appear to be truly superior transducers, except for a 6 dB peak at 15 kHz when the tweeter is in the head-on (axial) position. A number of listeners complained about an irritating quality on the top end when the speakers were played broadside; this may very well have been the cause—and one more reason to angle the speakers inward if space permits. On the bottom we found the useful response of the One/A to extend smoothly to approximately 40 Hz, rather than 32 Hz as claimed by Koss, but then a dipole is notoriously hard to measure at the lower frequencies. (As a matter of fact, there exists no proven mathematical model for low-frequency propagation by a large, free-standing, planar transducer.) The subjective quality of the bass is by and large excellent: deep, firm and authoritative on program material that contains genuine LF information. Except . . .

That peculiar thickness or muffled quality kept bothering us, without much of a clue as to where to look for it. We finally nailed it—we think. At exactly 50 Hz, the entire frame of the speaker system takes off, creating a completely uncontrolled resonance and inducing the panels to resonate as well. The resonant band is quite narrow (high Q), so that it isn't obviously excited by typical program material, making it somewhat elusive without careful sweep testing. Once found, the amplitude of the resonance turns out to be somethin' else: 8 dB for the integrated response of the four bass panels, 20 dB at the surface of the individual panels. One bass panel we measured also resonated at 57 to 59 Hz; this may have been a problem within the panel itself, but the overall 50 Hz resonance appeared to be definitely a frame problem. With two strong men squeezing the frame together from both sides, the center frequency of the resonance was shifted by a few Hz without curing the problem. In our opinion, it would take a complete redesign, with much heavier bracing and generally sturdier construction, to eliminate the resonance. And we're pretty well convinced that the coloration we heard was caused by transient effects at the 100 and 150 Hz harmonics of the 50 Hz fundamental. Without this defect we'd be inclined to rate the Koss Model One/A just a hair below the Beveridge Model 2SW, since it handles power a lot more comfortably than the latter even if it doesn't quite have the same super transparency. The way it was, many listeners preferred not only the Beveridge but even the DCM, which sounded somehow more spacious, open and natural in the all-important lower midrange.

The One/A also has separate terminals in the back, just like the Infinity QLS, to allow biamping the bass panels and the rest of the system separately if you so desire. The difference is that a convenient toggle switch is provided for instant change from the single-amp to the biamped mode. That made it very easy for us to determine that the difference (with one Bryston 4B vs. two Bryston 4B's) was barely perceptible without an electronic crossover. The biamped mode was declared very slightly more open and transparent by all listeners after considerable agonizing. All our other remarks on the same subject in the Infinity QLS review above apply equally here.

We recommend the Koss Model One/A, then, to all those who can't live without the clarity, inner detail and generally unsmeared sound of a full-range electrostatic, but at the same time won't spend the extra kilobucks to go all the way to the Beveridge or to something even more elaborate like the HQD System (Mark Levinson's creation for oil sheiks). On the other hand, for a lot less money, a pair of Rogers LS3/5A's with or without subwoofers, or a pair of DCM Time Windows, will provide comparable (in some ways even superior) musical satisfaction.

# Ohm F

*Ohm Acoustics Corp., 241 Taaffe Place, Brooklyn, NY 11205. Ohm F "coherent-sound" speaker system, $1200 the pair. Five-year warranty, excluding cabinet; customer pays all freight. Tested #32053 (cabinet #19297) and #32068 (cabinet #19296), selected at random in sealed cartons from manufacturer's stock.*

*Editor's Note:* **To prevent rumors, anonymous letters, crank calls, slanderous know-it-all comments, and other manifestations of ugliness and paranoia endemic to the high-end audio scene, the following disclosure must be made. The Editor/Publisher was one of the original founders of Ohm Acoustics in January 1971 and for several years thereafter its largest single stockholder. His involvement in the company was strictly as a Director and nonresident consultant, without any day-to-day participation in management or engineering. In March 1974 he sold all of his stock back to the company and has currently no closer ties to Ohm Acoustics (or any other audio manufacturer or retailer) than to the Exxon Company or the government of Liechtenstein. He does confess, however, to continued partiality to the brilliant, if incomplete, loudspeaker theory of the late Lincoln Walsh.**

The Ohm F is the great paradox of this speaker survey. Its ability to reproduce single pulses of different widths (in our tests between 1 msec and 0.1 msec) could conceivably be judged second to none, perhaps not even to the Beveridge. Its frequency response is also quite acceptable, except for a few anomalies noted below. Yet its sound is more highly colored ("canned" may be the better word) and less musical than that of any other speaker in this admittedly formidable group. Whenever we switched to it in the middle of a listening test, the instant reaction of those present was, "Huh? What on earth is *that*?" The midrange, especially, sounded like a tin can next to the really accurate speakers.

Why? We think we know exactly why. The Ohm F rings. It rings more than other speakers. It rings like a telephone. It isn't even possible to localize the ringing at specific frequencies; our test sample rang virtually everywhere. At 11 kHz, where there was also a 6 dB peak in the amplitude response, our sample did worse things to a tone burst than we had ever seen in our laboratory. But then 11 kHz isn't in the most sensitive range of the ear. Maybe it was the 900 Hz ringing that was the real culprit. Maybe some other frequency. There were just too many bad spots to choose from. On top of it, the frame of the Walsh driver resonated at 118 Hz.

Interestingly enough, it seems that another investigator has come independently to the same conclusion. Long after our own tests, we discovered that the DCM 'Time Window' brochure uses the Ohm F as the classic example of ringing. Not by name, of course; but the oscilloscope photograph they use as their example is identical to the one labeled Ohm F in their fifteen-speaker comparative pulse test further below on the same page. The photograph also shows that the Ohm F reproduces the initial pulse perfectly, exactly as we observed, but then just keeps on producing an output without an input.

We don't want to create the impression that this defect of the Ohm F is a vulgar design error, such as we occasionally pillory in this publication. It's damn hard to make a Walsh driver that doesn't ring. The basic requirements of the cone present a conflict. Since the Walsh speaker has been around for more than five years, here's just a quick reminder of how it works: A single cone, mounted convex side out and apex up, acts as a transmission line (in the antenna sense, not the bastardized acoustical labyrinth sense). Sound travels down the side of the cone faster than in air, and through proper choice of geometry the horizontal component of the traveling wave synthesizes a coherent cylindrical wave front in the air, starting at the cone surround. That's the theory, anyway; its divergences from real life belong in another discussion. In any event, the cone material must be extremely stiff (i.e., have a high Young's modulus) in order to have the proper sound propagation velocity. That means a high Q. At the same time, the cone material mustn't ring; it must be well damped. That means a low Q. Thus the ideal cone material for the Walsh driver would combine the most desirable characteristics of beryllium and mucus. No wonder that Ohm hasn't found it yet. They try to get away with a mundane combination of titanium, aluminum and paper in tandem, which doesn't quite make it. As we've said in the past, the Walsh invention would have deserved a multimillion-dollar R and D program to solve these and other immensely difficult design problems it poses. In which case it might have become the world's simplest and most accurate speaker. It's far from that right now.

A less serious objection we have to the Ohm F is that somewhere around 36 or 37 Hz, its response in the sealed enclosure has a 3.5 dB hump, meaning a system Q of roughly 1.4. That's too loose and boomy in our opinion; it represents the rock-pop taste rather than the devotion to accuracy one would expect of the knights of Walsh. (Unless, of course, the speaker is put plunk in the middle of the room, in which case the Q is approximately correct—but how many people will use it that way?) The -3 dB point is about 28 Hz, but that's 6.5 dB below the hump, so you don't get the subjective impression of a flat 28-Hz box. On the contrary, in nearly all room positions there remains some rather annoying midbass boom.

So, if you want one of the most brilliantly conceived speakers of all time—with a boomy bottom; a highly colored, ringing, metallic midrange and top end; plus superb time response if your ear could only separate it from all that mess—get the Ohm F. Incidentally, this is the "improved" version; some minor changes were made about a year and a half or two years ago. We remember the original version as having been better, but we didn't have all these other speakers to compare it to, and besides we don't claim an acoustic memory quite that retentive and precise. It's possible that our standards of excellence have changed more than the speaker.

# Pyramid 'Metronome'

*Pyramid Loudspeaker Corp., 71-07 Woodside Avenue, Woodside, NY 11377. Metronome Model 2 + 2W speaker system, $2525 the pair (in black formica, as tested; less in other finishes). Three-year warranty. Tested #169/269 and #170/270, on loan from manufacturer.*

In our Summer CES issue we gave a quick preview of Dick Sequerra's latest all-out attempt at SOTA and reported a favorable first impression of the Metronome speaker system. Just how the CES units resembled, or differed from, our test samples we'll never know for sure; until very recently Dick was going through a modification-of-the-week phase, so the only fair way to structure this review is to report everything in chronological sequence.

When our test samples first arrived, we were assured that they were typical of the limited number of units already sold and then being produced. They turned out to be absolutely tops in our survey in two respects: dynamic range capability and left/right imaging (among the larger units, that is; the minisystems are inherently excellent imagers). The Metronome was able to produce mind-blowing SPL peaks with suitable amplifier power behind it; it just couldn't be made to burp, grunt, spit, sizzle or show any other sign of distress no matter how hard we drove it. Clean as a whistle. And left sounded like left even when we moved all the way to the right. Just like real life, you say? No, the Metronome didn't sound like real life at all. The top end wasn't really open; the inner detail had no lifelike clarity and delicacy; everything was a little thick and smeared. We were both intrigued and disappointed.

In the laboratory, this particular version of the Metronome turned out to be almost perfectly flat in frequency response up to 15 kHz, at which point it cut off rather sharply. The only significant departure from a dead-flat profile was a 6 dB hump at 60 Hz, a little bit peculiar in view of the 37-Hz system resonance specified for the Model 2W subwoofer in the Pyramid literature. It seems that the response we measured, indicating a system Q of 2, was synthesized by a combination of the subwoofer, its roll-off network, and the lack of low-frequency roll-off on the Model 2 top section that links up with the Model 2W. (The Model 2 is allowed to run wide open; it's a complete and listenable three-way system all by itself, with a somewhat overdamped 8-inch woofer.) Despite the 60-Hz hump, the subwoofer didn't sound boomy, just rolled off. No wonder: the -3 dB point was at 30 Hz, meaning that the response dropped 9 dB in the all-important octave from 60 Hz to 30 Hz. According to Dick Sequerra, this is a deliberate design feature to avoid subsonic garbage and acoustic feedback at the high SPL's the speaker is capable of. We aren't sold on this philosophy, as you can probably imagine; nor do we believe that a 14-inch driver is correct for the internal volume of the subwoofer enclosure, which isn't much larger than that of the biggest bookshelf speakers. This is just the opposite of the Infinity QLS woofer approach: an oversize driver in an undersize enclosure. At least the oversize driver is never asked to move more air than it's able to, hence its apparent freedom from the usual eructations. But it can't give you the *low* lows, as it could with more volume behind it. All in all, though, this was one hell of a flat speaker system.

The speaker was also quite free from ringing throughout its range, but its problems became apparent when—you guessed it—we tested it with pulses. This particular version reproduced a 1 msec pulse satisfactorily, got progressively worse as we narrowed the pulse to 0.4 msec, completely reversed the polarity of a 0.26 msec pulse, and was totally unable to reproduce a 0.1 msec pulse. A classic case of time smear. We were then told by Dick Sequerra that the 2-inch tweeter was undoubtedly wired into the system out of phase. We had him reverse the phase, after which the impulse response got worse. Dick then decided that the original wiring had been correct after all and reversed the phase back again. So much for version one.

Sometime later, we received a call from Dick informing us of a new 4-inch midrange driver that was supposed to improve the performance of the Metronome considerably. Our Model 2 top sections were taken away to be modified and returned to us with the new midrange installed. The results were unbelievably bad. The beautifully flat frequency response was gone; the midrange output was elevated and appeared to have extreme phase incoherency (180° out with respect to the rest of the system); impulse response was the worst yet—and the sound was definitely degraded, with unmistakable time smear. A few days later we got another phone call, apologizing for the horrible mistake; the midrange had been incorrectly modified and would we please return our top sections once more for the correct modification. End of version two.

Version three arrived shortly thereafter, with assurances that this was It—the final, irrevocable production model. All units already in the field would be modified free of charge to conform to this one, we were told. And sure enough, this new version sounded different and, to our ears, somewhat better. More open highs, inner detail more clearly etched, more air around the instruments and voices. All without compromising the unique dynamic range and imaging capabilities. We still preferred the Beveridge by a wide margin and the Rogers as well as the DCM by a narrower margin; they just sounded more real, with more of a see-through quality, even though the Metronome was objectively the "cleanest" in the sense that it produced no grit or crud, no matter how complex or dynamic the program material happened to be. At the same time, something was subtly wrong with this new version that we hadn't noticed with version one; the latter had been less transparent and real but somehow more consistent, stable and balanced in sound quality. We weren't sure what the difference could be attributed to.

We took the speaker to the lab and realized almost immediately what was going on. For the first time, the Metronome was reasonably accurate in the time domain, with vastly improved results on the pulse tests, though still not in the Beveridge category. At the same time, the frequency response was completely shot to hell. So much so that it would be futile to try to describe its dips and bumps, since it was totally unrelated to any discernible zero axis. As a desperate example, there appeared to be a 10 dB peak at 6 kHz. With respect to 0 dB at what frequency? Don't ask.

What this proves is that good time response without good frequency response sounds more like real life than good frequency response without good time response. What it doesn't prove is that the Metronome speaker system is a finished product. We know Dick Sequerra well enough to predict that he won't be satisfied very long with the current production model. He is a perfectionist and wants basically what we want. He has accomplished a great deal already: reasonable size for a super system, good looks (we love that 45-inch tall metronome), fantastic construction (mostly 1½-inch walls!), stupendous dynamics, no gritty garbage out with no garbage in, precise stereo imaging, and now a fair degree of freedom from time smear. But that's not enough. For two and a half grand, we expect more from the man who gave us the Sequerra Model 1 tuner than a 50 Hz to 15 kHz speaker system that doesn't sound quite as real as the little Rogers at one fifth the price.

# Rogers LS3/5A

*Reference Monitor International, Inc., Suite 309, 4901 Morena Boulevard, San Diego, CA 92117. Rogers LS3/5A BBC Monitor Loudspeaker, $450 the pair (approx. $250 if purchased in Great Britain). Five-year warranty. Tested #S01777A/B and #S01807A/B, on loan from distributor.*

The Rogers LS3/5A, a British import, was originally developed for the BBC as a very compact monitor. Its outside dimensions are 12 by 7½ by 6½ inches. And it sounds better, at least from about 120 Hz on up at moderate levels, than any other speaker in this survey, except the Beveridge and the Koss—and we aren't so sure about the latter (see our Koss review above). For example, it's distinctly superior to the Infinity QLS.

How can a little shoebox do that? Well, how did a few Spitfires and Hurricanes beat the Luftwaffe? (And, rest assured, Hermann Goering was a harder nut to crack than Arnie Nudell.)

Actually, old chap, it's bloody simple. You take the 4-inch KEF bass driver and the ¾-inch KEF dome tweeter. They're jolly good, don't you know. You cross them over with a proper network made of decent parts (it's no place to skimp by using cheap electrolytics, etc.). You pay special attention to the time and phase characteristics. You make the sealed box as solid and rigid as Westminster Abbey. And then you cheat a little.

By cheating we mean the bass response profile. It's a bit of a humbug: would you believe +6 dB at roughly 130 Hz? Now that's a Q of approximately 2, and all the standard texts will tell you that the -3 dB point will then be around 75 Hz, which works out just about true to life. In other words, if the hill is high enough, its foothills won't drop out of sight, either. Is this any way to get bass response? We don't particularly like it; our strict upbringing makes us recoil from it—but how else are you going to extract a subjective impression of bass out of that itsy-bitsy box? Most people are fooled; time and again visitors told us that the bass sounded just great—until we showed them real bass. In all other ways, however, the Rogers has impressively flat frequency response, all the way up to the limits of hearing. By 20 kHz it's down somewhat, but then the English have always been kind to their dogs.

Our pulse tests revealed outstanding time-domain behavior and also explained why the Rogers sounds best with the tweeter end of the box tilted backward or angled inward. The on-axis impulse response showed some anomalies, but 45° or so off axis the two drivers fell into a time-aligned configuration, rendering better pulses between 1 msec and 0.1 msec than any other speaker tested so far, except the Beveridge. In view of the latter's size and weight, the Rogers became our convenient laboratory reference standard for pulse shape and coherence. Even so, it did exhibit a slight negative blip from the tweeter at the leading edge of a positive-going 0.1 msec pulse; however, all other conventional speakers, even the DCM, were doing the same thing to a considerably greater extent. Furthermore, the Rogers was remarkably free from ringing in comparison with the others. It's a delightful or scary little bugger, depending on whether you're an audio enthusiast or a speaker manufacturer.

A pair of LS3/5A's, separated by the right distance for your listening position and properly angled, will give you beautifully focused, crystal-clear, unsmeared, highly detailed sound reproduction, with all the spatial information (front/back, left/right) you could ask for. No reservations or qualifiers necessary. The Rogers will also resolve differences between amplifiers with ease and therefore deserves a very good one. The Norwegian *Electrocompaniet* power amplifier appears to be a particularly happy choice, since it can't possibly develop enough power across the speaker's 15-ohm impedance to hurt it (the Rogers is rated for 25 watts speech and music—not continuous power) and also happens to make the speaker sound superb. (We blew out the woofer on one of our test samples with the equally great-sounding zillion-watt Bryston 4B; don't *you* be that stupid.)

All right, what about that bass? Some people have suggested using two pairs of LS3/5A's to extend it, but that's not the way Mother Nature works. All you accomplish is a 3 dB increase in level (and power handling) with basically the same response profile. And you screw up the time response with the multiple sources. Our suggestion: live with one pair, as is, or go to a biamped system with subwoofers. We found the Fundamental Research to marry very well with the Rogers, as long as the nominal crossover frequency was kept in the 100 to 120 Hz region. Ideally, the crossover should be a little higher, say 200 or 250 Hz, to get completely away from the bumpy area of the Rogers; however, the Fundamental Research has some problems only an octave or so above that range (see our FR review above), so we had to stay lower. A properly crossed-over, biamped Rogers system with subwoofers promises to be one of the best money can buy; we plan to go more deeply into the subject and make more specific recommendations.

Meanwhile, we suggest you carry a pair of Rogers LS3/5A's around with you (in your car trunk, in an overnight bag, or whatever) when you go hi-fi visiting. After listening to your friend's (or dealer's) $2500 speaker system, you connect your little shoeboxes and you've got an enemy for life.

# Snell Acoustics Type A

*Snell Acoustics, 10 Prince Place, Newburyport, MA 01950. Type A loudspeaker system, $1370 the pair. Five-year warranty; manufacturer pays return freight. Tested samples on loan from manufacturer.*

This is a unique speaker system, in more ways than one. To begin with, it's by far the best-looking *large* speaker we can remember: an utterly simple, virtually seamless, upright brick of polished wood and stretched cloth, four feet high, two feet wide and just over one foot deep, designed to stand flat against the wall. The cabinetwork and finish are of a quality hardly ever seen in the hi-fi business; furthermore, the unit separates into a lower section (housing the downward-facing 10-inch woofer) and an upper section (with 4-inch midrange and 1-inch tweeter), so that two not very strong people, or even one in a pinch, can move it around without breathing hard. Just beautiful—though admittedly not the most important thing.

Well, how important is frequency response? All right, listen to this. Nearfield response of the individual drivers in their passband is dead flat (and we mean plus or minus close-to-nothing, right on the zero line) from 38 Hz to 22 kHz. The bottom end rolls off to -3 dB at 28 Hz, -10 dB at 20 Hz, -20 dB at 12 Hz. On top, the response dips to -6 dB at 24 kHz. So, using the convention of -3 dB corner frequencies, the total passband can be called 28 Hz to 23 kHz—and no ripples. How about that? We've never seen anything like it. What's more, all drivers radiate this output without significant peaks or dips over a very wide angle, so that with the crossover used the power response into the room is also astonishingly uniform. Distortion is low; power handling is very good; the woofer is particularly impressive—the driver and the enclosure seem to be exactly right for each other. (All you Q = 2 guys, please note.)

So how come the Snell isn't our top choice in this survey? You already know the answer if you've read this far in our alphabetical sequence of reviews: the frequency domain isn't where it's at. Mind you, the Snell sounds excellent. If we hadn't been exposed to the Beveridge, the Koss, the Rogers and the DCM, we would have lived with this kind of solid, wide-open sound quite happily for a long time. It's just a wee bit on the bright side (probably on account of the uniform power response right up to the highest frequencies), but we don't really mind that. Besides, the tweeter output can be reduced at the flick of a toggle switch. And the overall sound is truly clean and balanced. But something is lacking. It's that utterly alive, high-ceilinged, see-through quality and unsmeared resolution of inner detail available only with more or less time-coherent systems. Which the Snell isn't.

In fact, the Snell can't reproduce single pulses at all. You feed in one pulse, and out come two discrete pulses—one from the midrange and one from the tweeter. The two simply aren't joinable, regardless of distance or angle. This is an inherently noncoherent system, probably on account of the large acoustic separation between drivers. Too bad, since it's obviously a system designed with tender loving care.

In a way we're glad, perhaps at the expense of Snell Acoustics, that this speaker is such a pure embodiment of the frequency-response-above-all approach. If that were indeed the correct design philosophy, there couldn't possibly exist a better-sounding speaker, and we'll soon hear from some highly vocal sources just how wrong we are and where we should go to have our ears examined. On the other hand, if we're right—if time response is the more important design consideration—then the Snell Acoustics Type A will have served as the classic control experiment. Either way, it's a far from negligible contribution to speaker design.

# Recommendations

Please remember that the 15 speakers reviewed here, plus the dozen or so other units we've commented on in the past, represent only a small fraction of the universe open to serious audio enthusiasts. We can't swallow that whole universe in one issue—or even in four or six. If you're looking for infallible and all-encompassing guidance, you've come to the wrong place. On the other hand, we know the stuff we've tested pretty well and can offer the following circumscribed recommendations with some degree of confidence.

**Best speaker system tested so far, regardless of price: Beveridge Model 2SW (with reservation about headroom—see review).**

**Close to the best at a much lower price: Rogers LS3/5A (but bass not quite adequate without expensive subwoofer—see review).**

**Best full-range system per dollar: DCM Time Window (supersedes Dahlquist DQ-10).**

# A Comparative Survey of Power Amplifiers: Part II

By the Staff of
The Audio Critic

In this second installment of our search for the perfect power amplifier, we come a lot closer. We even nominate a new reference standard (at least until the next go-around) and also have other good news to impart.

As we explained in the introduction to Part I, we're convinced that a correlation can be established between the audible and measurable characteristics of a power amplifier, although the road that leads there is a rocky and winding one. Since this survey will have a concluding Part III in the next issue, we'll wait until then (as we said we might) to report just how far down that road we have, or haven't, been able to travel. We're still very much in the process of sorting out the tentative correspondences we've found so far between our listening-room and laboratory observations, and we certainly don't want to be guilty of premature or superficial generalizations, despite the blandishments of some fairly juicy data. One thing is certain: the Hirsch-Houck type of testing is insufficient and leads only to the discovery of one SOTA amplifier after another. We're looking for just the opposite: a meaningful test or series of tests that no amplifier in the world can pass without some difficulty. That way progress will be measurable; right now, if you read the commercial magazine reviews, perfection is being routinely attained.

Just so we won't leave you in total suspense, we're willing to confess to continued partiality to the CCIF intermodulation distortion test, with 14 kHz and 15 kHz mixed 1:1. We aren't quite sure, however, just how fine-tuned a tool it is; gross differences in the resulting sidebands from amplifier to amplifier invariably reveal audible differences; differences of a few dB may be imperceptible. We've also noticed interesting correlations between phase shift and sound quality, even though the resulting time delay at the frequencies in question may not, on the face of it, appear to be significant. And so forth and so on: we're knee deep in clues, without firm conclusions. Please be patient; give us another couple of months to find what no one else has in thirty years—or to report with some certitude that we're stuck.

**The continuing framework.**

Everything we've said about our listening test conditions in Part I still applies to the reviews below, but this time there's more. To wit:

In addition to the Mark Levinson ML-1, we also used the Hegeman HPR/CU (prototype) as a reference preamplifier. The latter is an outstanding unit, but please, don't jump to any conclusions. We haven't tested the two against each other, nor the Hegeman against the Rappaport, nor any of these against other preamps with SOTA claims that have recently appeared on the scene. We have no definitive opinion on the subject. As you can see on our back page, the question will be resolved in a new round of preamp tests to be reported in the next issue. Needless to say, we never changed two links in our reference chain at the same time. Whenever we compared two different power amplifiers, the preamp remained the same, whether it was the Mark Levinson or the Hegeman.

We also used other speaker systems this time in addition to the Dahlquist DQ-10. These included the Koss Model One/A electrostatic (a difficult test load for some amplifiers), the 15-ohm Rogers LS3/5A (used full-range as well as with the Fundamental Research subwoofer), and the DCM Time Window. Again, any two amplifiers under comparison were always listened to through the same speaker, although the more critical comparisons were repeated with several speakers.

We keep mentioning comparisons of two amplifiers, one against the other, since this new crop of power amps did require some careful A-B-ing to shake down the finalists. No wonder; quite a few of them were superior to our previous top recommendation.

Otherwise, our reference system remained the same, except that the Denon DL-103S (played through the Verion transformer) permanently replaced the EMT; also, the modified Stellavox 'Stellamaster' tape deck introduced in our speaker survey in this issue was used extensively with the master tapes mentioned there.

On to the reviews, then.

# Bryston 4B

*Bryston Manufacturing Limited, 17 Canso Road, Unit 1, Rexdale, Ont., Canada M9W 4M1. Model 4B basic stereo power amplifier, $1250. Three-year warranty; manufacturer pays return freight. Tested #4139 and #4140, owned by The Audio Critic.*

This is it, music lovers and audio freaks—the best power amplifier we've been able to find so far. Of course, we haven't tested them all. (Two that we should definitely look into, and haven't yet, are the Electro Research and the Mark Levinson ML-2. And there are others.) But this will be the one to beat, from now on.

The Bryston 4B is actually two separate mono amplifiers on one chassis. The only thing the left and right amplifiers have in common is the line cord. Each channel will deliver up to approximately 40 volts across just about any load that doesn't approach a short circuit—meaning 200 watts into 8 ohms, 400 watts into 4 ohms, 500 watts into 3.2 ohms, you name it. Since it's a truly unflappable voltage source with a brute-force power supply that never seems to run out of juice, the amplifier simply doesn't care whether it sees an obscenely complex load like some of the full-range electrostatics or a virtually pure resistance like a Cizek box speaker. It just keeps rollin' along. That's probably the main secret of its listening quality, but since its designer has expressed the engineering philosophy behind it in his own words, which we're reproducing herewith, we'll refrain from second-guessing him. We can't resist wondering aloud, though, why there's all the fuss about pure class A output stages both in this country and in Japan, when in Canada class AB can be made to sound this good with this kind of power.

Because the Bryston 4B sounds just about unexceptionable to our ears. On the top end it rivals the *Electrocompaniet* amplifier, which is our standard of excellence on that count—open, sharply etched, unsmeared, fast and sweet at the same time. And of course the 4B eats the

# Comes The Resolution, All Power Amps Will Sound Great

*Editor's Note: Whenever we're sufficiently impressed by the performance of an audio component, we try to brain-pick its designer for some kind of correlation between engineering philosophy and sound quality. Having been thus impressed by the Bryston 4B power amp, we managed to elicit the following observations from Chris Russell, a very serious and very long-haired young man who not only designed the circuit but also, with his brother and father, runs the company.*

The Audio Critic:

In an era when power amplifiers are attaining distortion measurements approaching a thousandth of one percent over a wide range of frequencies and output levels, essentially falling to background noise levels which are theoretically inaudible, why is there the continuing search for an amplifier which is entirely accurate to the ear musically? Whence springs the departure from theory which seemingly separates objective and subjective realities? Is it possible we are hearing things we cannot (or do not) measure? Let us investigate.

The latest thinking on the problem of dynamic distortion mechanisms in amplifiers centers largely on the popularization of Transient Intermodulation Distortion. As it is known, this is a feedback-defeating phenomenon related to the ultimate speed or slew-rate capability of the amplifier. Feed an amplifier circuit a transient signal faster than it can handle and it generates a momentary burst of distortion trying to catch up.

It should therefore be possible to "perfect" an amplifier by either low-pass filtering the input or pushing the slew-rate limit to stratospheric values. That this engenders only *some* improvement leads us to the inescapable conclusion that it is a gross oversimplification of a very real problem. It is even possible that exclusive concentration on this one particular aspect of an amplifier's behavior can actually have detrimental consequences on overall musical reproduction.

In point of fact, there are dozens of phenomena which can encroach upon an amplifier's ability to handle a musical signal with negligible distortion. Most are transient-related. Many are *low*-frequency problems. For instance, when an amplifier delivers a pulse of low-frequency energy to a loudspeaker, the current flow tends to pull down the power supply voltage. This change in voltage supply throws all the sensitive standing currents and bias voltages within the amplifier into a state of flux, muddying the signal. Tying both channels to the same power supply exacerbates the problem greatly, as then one channel can affect the other. Split up the power supplies and regulate the voltage to all the sensitive stages, and there is an immediate and obvious improvement in clarity, transparency and resolution.

An interesting word, that—resolution—for it implies much more than appears at first glance. Resolution denotes focus, the ability to delineate subtle and delicate details in the musical picture, and its absence can have as much to do with the removal of something as with an addition such as harmonic distortion. It is entirely possible for a circuit to "lose" subtle information, or at least to hide it behind the coarser fabric of the overall musical framework.

It appears to be this loss of resolution which offends the ear rather grievously in many circuits which display a certain amount of nonlinearity in the signal processing, and which hope to straighten out the problem with feedback or with "complementary nonlinearities" further down the line.

Case in point: Another low-frequency vagary which often creeps into amplifier circuits is related to high standing currents in class A stages. When a transistor conducts a good deal of current, at a rather high voltage, it heats up—since voltage times current equals power. When the signal causes the voltage to swing up and down, the power dissipation varies from top to bottom of the cycle. At mid to high frequencies this tends to integrate over the cycle, but at lower frequencies the transistor chip will have time to vary as much as tens of degrees Celsius from one end of the signal swing to the other. Since most of the transistor's characteristics, including gain, base-emitter bias voltage and frequency response, vary with temperature, this causes nonlinear distortion. Further, much of it is high-order. Common sense dictates, therefore, that we should run low-level class A stages at no more current than necessary, and at reasonably low voltages if possible. (Especially in view of the fact that, other things being equal, transistors distort less in every way when they handle less current.) This way the junction will always be near room temperature and not subject to change with signal swing.

The one unfortunate fly in the ointment is that transistors tend to increase in frequency response with increased current flow. Thus the Great Slew-Rate Race dictates that many designers opt for speed rather than linearity and ignore this common-sense ideal. (Recently I have seen some amplifiers with low level stages dissipating so much current that they require small power devices on their own heat sinks, running blazing hot to the touch!) Since amplifiers tend to have gobs of feedback available at low frequencies, this problem is often hidden from steady-state distortion measurements, but it is still audible as a loss of resolution. Unfortunately, once the finesse in separating tiny nuances is sacrificed, it can never be regained later.

This is why I say that, to the ear, distortion is distortion. There is no point in taking a step backwards in one area in order to gain in another. Speed and slew rate are very important parameters, but only insofar as they relate to obtaining the maximum benefit from a circuit which is maximally linear at all frequencies, as well as from the amount of feedback which is employed. (After all, feedback can and does further reduce distortion.) The important watchword is overall musical accuracy.

Thank you very much for your interest.

Sincerely,
Christopher W. Russell
Bryston Manufacturing Ltd.

Electro for breakfast in overall drive capability and load handling. The Bryston also has the best bass we've heard so far, free from the least trace of looseness, hangover or artificial "warmth." It just goes down, down, down, tight and unshakable. The midrange may not have the uncanny clarity of the Futterman, but it's close (damn close), and the overall quality of the Bryston is more open than that of the Futterman. Actually, as a high-power amplifier for all seasons, the Threshold 800A was the one the Bryston had to beat, which it did quite handily. By comparison, the Threshold, a very good-sounding amplifier in its own right, seemed a bit heavier, thicker and more closed-down in its sonic presentation, with just a least bit of the aforementioned "warmth" factor.

Incidentally, the above observations apply only to a Bryston 4B that has been turned on for half an hour to an hour. Listened to cold, the amplifier sounds a little hard and glary. The warm-up seems to stabilize the circuits.

In the laboratory, the Bryston 4B appeared to be irreproachable, with point-double-oh or low single-oh distortion on every possible THD and IM test. No problem at all on the 14-plus-15-kHz killer test; gorgeous square waves right down to 20 Hz and below; in fact the phase shift at 20 Hz is an incredible 1° (don't forget, this is not a DC amplifier) and totally unmeasurable by the time you get up to 100 Hz. The slew rate appeared to be a little over 40 V/uS, which is somewhat short of the "greater than 60 V/uS" specified, but then who is to say that Bryston's technique for this rather difficult measurement isn't superior to ours. In any event, you can double the slew rate by throwing the two channels into a bridged mono configuration at the flick of a switch (what a nice feature!), in which mode the available power is 800 watts into 8 ohms. That's more than one horsepower.

As a footnote to all this, we must refer you back to *The Admonitor* column in our second issue, where we took Bryston to task for claiming to have the world's best power amplifier, period. We still stand firm on that admonishment, even though the claim seems less outrageous after our tests. When an ad makes a superlative product appear like a hype, it's time to switch to another ad writer.

Oh yes. What about the Quatre DG-250, our previous top choice? Not quite in the same class with any of the amplifiers mentioned above.

# CM 914

*Audio International, Inc., 3 Cole Place, Danbury, CT 06810. CM 914 Stereo Power Amplifier, $449. Tested #3129, on loan from manufacturer.*

Since the Quatre DG-250 seems to be leaving a very poor track record as regards reliability, this neat little 100-watt-per-channel amplifier from the successor of C/M Laboratories is the best we've been able to find in the same price range as an all-around workhorse alternative. The CM 914 sounds more open than the Quatre on top, although we still prefer the sound of the latter on an overall basis. The CM 914 is a little less focused and transparent in the midrange and not as firm in the bass. The very fact, however, that the CM can give the Quatre a hard time in an A-B comparison shows that it's a better amplifier than most of those reviewed in Part I, including the bigger and more expensive CM 912a. (It's interesting to note that the 912a incorporates current limiting whereas the 914 does not.)

Our laboratory tests revealed some peculiar ringing on square waves, which was sometimes intermittent, sometimes stubborn and ineradicable. Since it existed in both channels, we doubt if it was due to some kind of minor defect in a single component. The ringing, excited by the leading edge of any square wave, appeared to have a frequency of approximately 1 MHz and a decay time of about 3 uS. Whether it had anything to do with what we heard . . . who knows?

Otherwise the amplifier exhibited single-oh distortion within its power range on our THD and IM tests, except on the 14-plus-15-kHz CCIF, where it did a little worse though still acceptably (better than the Threshold 800A for example). Its strongest spec seems to be phase shift; there's hardly any to speak of between 20 Hz and 20 kHz.

As audio purists, we can't get too excited over this amplifier; on the other hand, we must admit that it represents decent performance and value when measured against what else is available.

# D B Systems DB-6

*D B Systems, PO Box 187, Jaffrey Center, NH 03454. DB-6 Precision Power Amplifier, $650. Five-year warranty; manufacturer pays return freight. Tested #200730, on loan from manufacturer.*

When a 40-watt-per-channel power amplifier of super-austere appearance costs $650 (i.e., considerably more than the much more powerful Quatre DG-250, Quad 405, CM 914 or GAS Son), one would expect it to produce extraordinarily accurate sound at moderate levels. The DB-6 sounds clean, sweet and non-fatiguing, but not accurate. Hence we must call it a disappointment.

Example: We were listening to an unaccompanied quartet of voices through the Bryston 4B. Each voice stood out in perfect relief, without the slightest tendency to cover up the other three. You could actually "hear" the spaces between the four singers. We switched to the DB-6. The sound blended into a pleasant, euphonious glop. It wasn't unmusical; if we hadn't known what was *really* on the record, we would have been satisfied. Strange amplifier.

It's interesting and instructive to compare the DB-6 with the *Electrocompaniet* amplifier, which has a somewhat lower power rating (it also costs less) but is otherwise similar in its techno-image: a cute little black box for the purist. The Electro has a slew rate of 100 V/uS; the DB-6 is much slower at 15 V/uS. The Electro does a little better on the 14-plus-15-kHz CCIF test, but not much. (Neither is in a class with the Bryston 4B, even at low power levels.) Both are uncomfortable with low-impedance loads that tax the current capability of the power supply. Does the Electro sound considerably more accurate (a la Bryston) because it's faster? Or is that only the by-product of the actual causative factor? Or is it purely incidental? These are some of the nagging questions we're trying to find answers to. (Lots of luck, fella, say some pretty smart engineers.)

One thing that isn't responsible for the shortcomings of the DB-6, for sure, is harmonic distortion. The specs say less than 0.005% from 20 Hz to 20 kHz, and it's the truth—we've verified it. Double-oh and triple-oh distortion figures seem to be the big thing at D B Systems, and we're very much beginning to wonder whether that isn't the root of the problem. Ultralow distortion can only be achieved with lots of feedback; lots of feedback will cause . . . there we go again.

Another thing we aren't sure about is the net effect of the subsonic filter in the DB-6 (12-dB-per-octave Butterworth with 15 Hz corner frequency). There has been some talk of *audible* group delay introduced by certain types of subsonic filters, but not such simple networks. At any rate, the resulting phase shift at 20 Hz is 64°, which means a delay of just under 9 msec at that frequency.

We wish we could either like the DB-6 a little more or else be able to explain a little better just where the design went wrong. Neither of which wishes can change our conclusion: too little for too much.

# Electrocompaniet
**(Part II)**

*Electrocompaniet, Toyengt. 14, Oslo 1, Norway. "The Two-Channel Audio Power Amplifier," $505 to private users, $344 to dealers, FOB Oslo. Tested unnumbered sample, on loan from owner.*

Since our original review, we've found out a great deal more about this lovely little amplifier. First of all, it *does* really "exist" from the consumer's point of view; you can write for it to the above address and, sooner or later, they'll send you one. (For all we know, by the time these words are in print, there'll be an American distributor; don't count on it, though.) Secondly, it's every bit as good after greater familiarity with it as we first reported. Thirdly, the 100-watt-per-channel version is still only in the laboratory phase, not in production.

Our second sample of the amplifier was a little different from the first physically, but not in sound. The highs are still SOTA; completely unsmeared, etched but edgeless, totally transparent. The rest of the range is equally good, as long as you don't ask the amplifier to drive loads it doesn't like at levels it can't handle. Specifically, complex loads such as the Koss Model One/A, which drops down to rather low impedance at some points in the audio range, distress the Electro to a surprising degree. Even at moderately loud listening levels, the Koss/Electro combination sounded hard and irritating. On the other hand, the 15-ohm Rogers LS3/5A makes the Electro very happy; at that impedance the current demands on the power supply are reduced and the sound is of the utmost purity.

Interestingly enough, the amplifier *will* deliver 14 volts or so into any load without actually clipping (i.e., 12.5 watts into 16 ohms, 25 watts into 8 ohms, 50 watts into 4 ohms); however, at 4 ohms the power supply has trouble providing the necessary current and high-amplitude signals begin to be envelope-modulated by the 60-Hz line frequency. That's what we obviously heard on the Koss. Otherwise the Electro behaved very nicely on the test bench; even the 14-plus-15-kHz CCIF test went well, though not brilliantly. So, we must ask again, is it the 100 V/uS slew rate with 1 MHz bandwidth that makes this circuit sound so good, or is that the symptom rather than the cause? Tune in the next time around and see if we can answer the question.

Incidentally, the Electro has a rather peculiar input with a 1000-ohm impedance and mustn't be driven from a high-impedance source such as a tube preamp. Input sensitivity for full output is only 0.35 volts. *Electrocompaniet* makes a matching preamp, which we haven't seen, that has a 10-ohm output impedance, just the right gain for the power amp, plus a 70-kHz low-pass filter, presumably to protect the power amp against transient overload. Interesting.

Our recommendation: think about your speaker load, your loudness requirements, and consider this power amplifier very carefully if you want close-to-SOTA sound with the aforementioned limitations at a reasonable price—reasonable even with freight and import duty added.

# Futterman H-3aa

**(Part II)**

*Futterman Electronics Lab, 200 West 72nd Street, New York, NY 10023. H-3aa vacuum-tube power amplifier (mono), $260; stereo pair, $520 (no longer made to order at this price; see below about availability). Tested #105 and #106, on loan from manufacturer.*

Our test samples arrived as promised (see Part I) and confirmed just about all our initial impressions. This is unquestionably one of the world's finest power amplifiers, but not an amplifier for all seasons—all applications, that is.

First of all, the Futterman doesn't like to see low impedances, such as are presented by the Koss Model One/A electrostatic, for example. It just isn't on its best behavior at 4 ohms; it can't deliver enough power, for one thing, and clips badly when pushed. Go just a wee bit higher and the impedance match improves; for example the DCM Time Window, which is still a fairly low-impedance speaker, never sounded better than when we connected it to the Futterman. At 8 ohms the situation is pretty well stabilized, and at 16 ohms the amplifier comes into its own; but how many 16-ohm speakers are there? The Rogers LS3/5A is rated at 15 ohms but shouldn't really be used with the Futterman, which can deliver about 150 watts into that impedance and blow out the rather fragile KEF drivers. (We tried the combination, anyway, and it sounded gorgeous—but the Rogers/*Electrocompaniet* duo was, if anything, even better.)

As regards specific sound quality, the Futterman seemed to have the most amazingly transparent and unblurred upper midrange (female voices, etc.) of any amplifier that has crossed our path; superb highs (but not quite as fast, etched and open as the *Electrocompaniet* or the Bryston 4B); excellent bass if used with extra output capacitors (more about that below), and a slightly compressed overall sound in comparison with the Bryston (our reference standard). In other words, better than any other amplifier in *some* respects, better than all but two or three others in *every* respect. It must be that at least one unidentified kind of distortion, the kind that dumps spurious products into the all-important upper midrange, has been more thoroughly eliminated in the Futterman than in any other amplifier.

In the laboratory, the Futterman was no match for the finest solid-state amplifiers, such as the Bryston 4B. All our usual tests came out a little worse, though none of them disturbingly so. It seems that a tube amplifier is a whole different species, and the same rules don't apply. Which may prove, in the end, that the rules are wrong, since it shouldn't make any difference to the electrons how we make them flow, as long as they flow just so.

We mentioned the output capacitors in the

Futterman, which in the standard model add up to 2400 microfarads per channel. Julius Futterman supplied us with an extra 4200-microfarad cluster for each amplifier, so that each of our speakers could be coupled to the tubes via 6600 microfarads. And it made a difference. The bass was drier, more solid, more realistic than with 2400 microfarads. Whether this has anything to do with phase shift and time delay, we have no idea. It certainly has nothing to do with amplitude response. Even the standard model is flat at 20 Hz. It may be that electrolytic capacitors have some peculiar distortion which is minimized by multiplication in parallel.

The main problem with the Futterman is where to get it. We've been informed that Julius Futterman has stopped taking orders from private individuals at the ridiculously low price listed above. He is now selling every amplifier he makes by hand to The Sign of the Golden Ear (Riverview, Michigan), the outfit that plans to market the amplifier in a commercially packaged version, which to the best of our knowledge doesn't exist yet. The price of this latter version was originally planned to be $989 for a stereo unit or stereo pair. That's all we know.

If it's any consolation to you, a Futterman amplifier was just as hard to get in 1956.

# Quatre DG-250
**(follow-up)**

*For manufacturer, price and other particulars, see original review in Part I.*

We regret to tell you that our original endorsement of this amplifier must now be heavily qualified. It still sounds as good as ever (though not quite as good as the top amplifiers in this second batch), but just too many reports of failure in use have reached us to be ascribed to normal attrition. Since for every failure reported to us there must be a good many others we'll never know about, the total number should logically be a substantial percentage of production, since not that many Quatre DG-250's have been made thus far.

And that's not all. The typical DG-250 failure seems to be particularly drastic and destructive. Here's the scenario we've been hearing over and over again: The predriver transistor fails and a huge amount of DC, representing a major portion of the rail voltage, appears across the output. Since the amplifier has no protective circuits to sense this type of failure (don't ask us why), the DC passes through the speaker load. The speaker either isn't fused or the fuse doesn't blow fast enough. The speaker goes "arrgh" and that's that. No more woofer—and maybe no more other things. Not nice at all. Inexcusable, in fact.

Out of just three Quatre DG-250's that have passed through our own hands, one of them did this. A second one conked out nondestructively. That's two out of three. And similar cases we personally know of add up to a solid two figures, not one. On top of it, we hear that Quatre has been dragging their feet in responding to the situation. New circuit boards to replace the burned-out ones take forever to receive, we hear; and we've had reports of noncooperation and evasiveness by the front office. In all fairness, we must remind you that we spend most of our time and energy on equipment tests and editorial work, not on double-checking the authenticity and accuracy of complaints. But there must be a fire of some sorts where there's so much smoke. And you should have seen the smoke that came out of our DG-250 . . .

So, if you buy a Quatre, you're on your own as far as we're concerned. Enjoy the good sound, but don't say you haven't been warned. The amplifier may easily turn out to be more expensive than you thought.

# RAM 512

*RAM Audio Systems, Inc., 17 Jansen Street, Danbury, CT 06810. RAM 512 Stereo Power Amplifier, $1150. Three-year warranty; manufacturer pays return freight. Tested #10547, on loan from manufacturer.*

In our CES report we made the statement that Richard A. Majestic, who is Mr. RAM, was originally the M of the former C/M Laboratories, which is now Audio International. That was an error. Dick was an employee at C/M, not a founding partner. We also made the statement that his RAM 512 power amplifier was an "all-out design." That was no error, except in the intended sense. The amplifier we tested turned out to be an all-out design all right—all out to make money by tweaking super specs (high power, double-oh THD, etc.) out of a basically flimsy box and selling it at an exorbitant price.

If you're getting the impression that we're a little upset and indignant about this, you're quite right. The price of the RAM 512, either retail or wholesale, is only a few dollars below that of the Bryston 4B, our new reference standard. And the comparison between the two amplifiers is a joke. Compare the power supplies (single vs. dual), the heat sinking (hardly any in the RAM, just convection cooling), the quality of parts, the general construction (the Bryston weighs 50% more)—we could go on. Without knowing the prices, you'd guess a cost ratio of about two to one. What the RAM gives you for your money is a pair of big, flashy dB meters plus peak-reading -20 dB LED's, and lots of protective devices to keep the tweaked circuitry from going up in smoke or blowing out your speakers (at least that's conscientious, unlike Quatre). Even so, the RAM can't give you the same maximum volts into 4 ohms as into 8 ohms, which the Bryston does without batting an eyelash; and even into 8 ohms the RAM gives you a little less power, being rated at 170/170 watts.

Of course, the most important comparison is just listening—and, again, it's no comparison. The RAM 512 sounds a little overbright, hard and glary, though not disturbingly so; it gives a nice feeling of depth, but doesn't have anywhere near (and we mean anywhere *near*) the transparency and definition of the Bryston (or the Futterman or the *Electrocompaniet*, for that matter, at much lower prices). If the RAM cost, say, somewhere in the seven hundreds, we'd be more charitably inclined toward it and would probably weigh its positive and negative qualities with greater deliberation. At $1150, however, it's put up or shut up.

On the test bench, the RAM 512 displayed a peculiar kink in both the leading and trailing edge of a square wave (any square wave), which is definitely not normal or acceptable. (Overbiased transistors?) Phase shift at the lower frequencies was also considerably higher than we're comfortable with, although it's inevitable with the AC input coupling employed and we don't claim (not yet, anyway) that this is an established criterion of audible quality. On top of it, the amplifier went into protection on both channels toward the end of our tests and we couldn't make it work again without looking into the cause—which, quite frankly, didn't interest us at that point.

Our conclusion: give us more amplifier for the money, RAM, or else take out those meters, give up a few watts, and cut the price in half.

# Threshold 800A

*Threshold Corporation, 1832 Tribute Road, Suite E, Sacramento, CA 95815. Model 800A class A stereo power amplifier, $2275 (discontinued). Two-year warranty; manufacturer pays all freight. Tested #770408, owned by The Audio Critic.*

Even though the Threshold 800A has been discontinued as a specific model, we consider it well worth reviewing, since this company is strongly committed to the patented active-bias class A circuitry it incorporates. A smaller and less expensive version, the Threshold 400A (a 100/100-watt power amplifier at a mere $1147) is still on the market; and a successor to the 800A, tentatively called the 8000 and built on two separate mono chassis, is soon to be released. (The 810 we had mentioned in our CES report was shelved in the prototype stage.)

The 800A was undoubtedly too expensive and impractical to make: a huge but beautiful black monster, with massive handles and two enormous meters calibrated in watts at 4 and 8 ohms, practically hand-wired throughout—a real status symbol. The rated power is 200 watts per channel into 8 ohms; a conventional (constant bias) class A amplifier of this power would have to be the size of a small refrigerator. The Threshold system of active or dynamic biasing reduces the power dissipation requirement at idle by a factor of four; hence the large but still viable dimensions of the 800A. A critique of the Threshold patent is beyond the scope of this review; we have, however, heard a few impeccably qualified technologists speak of it with a certain degree of disdain. Their comments ranged from "class A in name only, not in fact" to "unnecessarily complicated without genuine advantages" and even to "class A is pure bullshit, anyway." On the other hand, the Mark Levinson ML-2 is a constant-bias class A design with some pretty good credentials behind it, so we'll hang fire on this one until we have an ironclad case one way or the other.

Since the proof of the circuit is in the sound, we must report that until the arrival of our two Bryston 4Bs, the Threshold was our reference amplifier for a number of weeks. It sounded more solid, sweet and open than the Quatre DG-250 (big deal, at about five times the price); it wasn't quite as uncannily transparent in the upper midrange and lower treble as the Futterman, but unlike the latter it could drive low impedances without any problem; nor did it have the totally unsmeared definition of the *Electrocompanient*, but with its vastly greater power capability it was a considerably more versatile and practical workhorse. We were quite pleased with it and listened to it with a great deal of satisfaction.

Then came the Bryston 4B and the pecking order changed somewhat. The 4B has a tighter, drier, more thoroughly lifelike bass; a more open, luminous quality throughout; and more snap and definition on top, without edginess. The Threshold sounds slightly thicker and darker; with sweeter but less finely delineated highs; somehow more closed down overall. Minor but noticeable differences, between two outstanding amplifiers. We've heard it said that the Threshold was trimmed in to make the Dayton Wright electrostatic sound as good as possible, since the two have the same channels of distribution and are generally demonstrated together. We hope that's not true; a power amplifier shouldn't be used as an equalizer or compensator; however, delivery of our Dayton Wrights was delayed (see the speaker survey in this issue), so we still don't know more than what we're told—which can be notoriously unreliable.

Our laboratory results were most interesting: the Threshold 800A did quite poorly on the 14-plus-15-kHz CCIF test, with 13 and 16 kHz sidebands averaging 12 to 15 dB above those produced by the Bryston 4B at the same power levels. This makes us wonder about the sonic correlatability of the test itself; after all, the Threshold does sound better than a lot of amplifiers that throw less obvious sidebands. It's the first significant exception so far, though. Just to follow through, we measured the standard (meaning SMPTE) IM distortion of each amplifier. Lo and behold, the Bryston was almost a whole order of magnitude better, with double-oh figures throughout as against the Threshold's single-ohs. How about that?

The Threshold also showed more phase shift at the lower frequencies than the Bryston (8° vs. 1° at 20 Hz, 2° vs. 0° at 100 Hz), but we can only repeat our reservation about the significance of this as stated in the RAM review above. In most other respects, such as slew rate and THD for example, the two amplifiers were quite comparable. The Threshold, however, doesn't have the Bryston's mono bridging feature that doubles the slew rate and quadruples the power.

Our suggestion, then, if you can afford the Threshold, is that you buy *two* Bryston 4B's, which won't cost you all that much more, throw the bridging switch and enjoy the fastest two 800-watt channels in the business. Even if faster doesn't mean better (which remains to be seen), it's nice to have that kind of headroom.

Unless, of course, you feel underprivileged without class A. In which case you might want to wait for the Threshold 8000—or get in touch with a Mark Levinson dealer. He'll show you what class A, and privilege, can do for you.

# Recommendations

Pending the conclusion of this survey in the next issue, which may still hold surprises, dark horses and even scandals, here are our current top choices.

**Best power amplifier so far, regardless of price: Bryston 4B.**

**Best sound per dollar: *Electrocompaniet* (if you can get one and are satisfied with 25/25-watt power capability).**

**Special situation for the connoisseur: Futterman H-3aa (*somebody* will get one next week or next month—why not you?).**

# Cartridge, Arm and Turntable vs. the Groove: Who's Winning?

## Part I: In which we emend and update our original article on tracking geometry, find a mind-blowing sonic gap between optimized alignment and currently accepted practice, and as a result send all equipment reviewers back to square one. Also: a common-sense critique of 10 tone arms and turntables.

In our first issue (January/February 1977, pp. 31-35), we published an article under the title of "Have Tone Arm Designers Forgotten Their High-School Geometry?" Our message (namely that correct geometry in a pivoted tone arm costs no more than incorrect geometry, yet there's no arm on the market that's 100% correct) put quite a few noses out of joint, for which the blame must be placed on the unfeeling laws of nature. If it's any consolation to those who were upset by our facts, we now consider that article to have been somewhat superficial. Not in the sense that the argument it presented was in any way incorrect; far from it. We still stand behind every statement made in it, as far as the physics and mathematics of the subject are concerned. But it's quite obvious to us today, after approximately nine additional months of investigation and listening, that we didn't go nearly far enough. We grievously underemphasized the sonic consequences of the mathematical sloppiness we were complaining about and gave no inkling of the chain reaction of invalid conclusions that can result from complacency about the cartridge/arm/turntable vs. groove relationship.

The man we must credit for steering us in the right direction and dispelling our initial doubts about the audibility of ridiculously small deviations from optimum tracking angle (both lateral and vertical) is Mitchell Cotter, the encyclopedically learned technologist of Verion Audio. Mitch has been spreading the word with apostolic zeal; he has undoubtedly aligned more record players just as a favor to the innocent and distributed more Xeroxed instruction sheets on how to do it than there are letters on this page. The Boston Audio Society is still reeling from his panzer assault on the subject (massively documented as usual) at their March 1977 meeting. Our own experiences since our first issue and recent conversations with other proselytes (who are still few and far between) bear out the Cotter doctrine without qualification.

**The unsettling but inexorable facts.**

It would probably be more logical to start with certain unarguable givens and then lead you step by step to our somewhat disturbing but nonnegotiable conclusions. But we feel that the latter are too important to be shared only with those who are willing to track with us through our exposition. Since we don't want to lose anybody at this juncture, not even tone arm designers with a short attention span, we've decided to spell out our message first and explain later. Thus:

**1.** Lateral tracking error, if large enough to be at all measurable with simple tools (straightedge, triangle, protractor, ruled paper, piece of thread, or whatever), is readily audible through a speaker/amplifier system of reasonably high resolution. Even the difference between the successive zero-error and maximum-error points of an optimally aligned pivoted arm is audible, let alone the difference between an optimally and a nonoptimally aligned arm.

**2.** Optimum lateral tracking alignment of commercially available pivoted arms can only be achieved by ignoring the manufacturer's instructions and twisting the cartridge in the headshell to obtain the correct combination of overhang and offset angle. All other approaches, including the one suggested in our first article, are unnecessary compromises.

**3.** Vertical tracking error (i.e., a difference between the original vertical cutting angle and the effective vertical playback angle) is similarly audible within the limits of the finest adjustments that can be made by hand (such as barely raising or lowering the vertical pivot of the arm, shimming up the record by means of a thin cardboard disc or another thin record, etc.). Without sophisticated test equipment, correct vertical tracking angle can only be determined by trial and error coupled with careful listening. Visual guidelines, such as the apparent forward tilt of the stylus, the angle between the stylus bar and the record, or the parallel position of the arm tube with respect to the record, are almost entirely useless.

**4.** Unfortunately, the correct vertical tracking angle is different for all records cut at different times, in different places, by different people, on different lathes, with different cutter heads. No adequately stringent industry-wide standard is in force, even today. This is very bad news, since it means that truly optimized playback requires constant adjustment by hand (and ear) from record to record, unless a uniformly cut series of records is being played.

**5.** What makes all the above agonizing worthwhile, or at least meaningful, is that optimum alignment of the cartridge and arm for both lateral and vertical tracking error improves the average sound quality of a record collection to an astonishing degree. Records previously thought to be hopelessly bad often end up sounding quite decent, and the better records begin to approximate what high fidelity is supposed to be all about: a creditable imitation of live music. In other words, phonograph records are a more accurate medium than jaded audiophiles suspect.

**6.** By the same token, listening tests and equipment reviews based on phonograph records are of highly questionable validity unless the lateral and vertical tracking alignments are religiously optimized and maintained. Since that's hardly ever the case, most evaluations of cartridges, tone arms, turntables and even other components in either the commercial or the underground audio publications can be expected to be quite erratic. Alignment according to the manufacturer's instructions will randomly favor or handicap A over B; after proper alignment the order of preference is often reversed. We now believe that this is one of the principal causes of disagreement among otherwise unbiased and keen-eared equipment reviewers as well as rank-and-file audiophiles.

This is admittedly pretty heavy stuff, and we don't expect it to penetrate some of the pointier heads in high places for some time to come. The consequences aren't exactly negligible, when you stop to think about them: all pivoted tone arms should be redesigned and their mounting instructions revised (and even then a correctly executed radial-tracking arm will be audibly better—but try to find one!); furthermore, all tone arms, whether pivoted or radial-tracking, should have a mechanism whereby the vertical tracking angle can be conveniently changed during play; or better yet, the recording industry should immediately adopt an ironclad vertical tracking angle standard and enforce it internationally. Are you beginning to see the dimensions of the problem? All we can accomplish here is to make a few small waves and hope that somebody out there gets splashed a little bit. Any kind of serious dialogue on the subject within the audio community would require the participation of some of the largest commercial interests, and Nick the Greek isn't putting heavy odds on that probability.

Meanwhile, we'll try to illuminate our main points with some simple explanations and also provide specific instructions for optimizing your cartridge/arm/turntable set-up. Remember, though, that this is only Part I of what we plan to be a continuing exploration of the subject, so that a number of questions may not be resolved to your satisfaction until next time.

**The stylus/groove interface.**

The key to understanding the requirements of distortion-free record reproduction is a clear and uncomplicated mental image of the stylus riding in the groove. All the facts and all the nonsense you've ever heard about cartridges, tone arms and turntables relate, more or less directly, to that single interface. If you think about just that and nothing else—if you don't get bogged down in peripheral considerations such as, say, bearing friction—everything falls into place and the subject is no longer open to techno-freak obfuscation.

Visualize the cutter stylus, plowing a stereo groove into a lacquer master. Its lateral motion is from side to side, exactly in line with the radius of the disc at the point of cutting. It's obvious, therefore, that the lateral motion of the *playback stylus* in the groove must be along that same radius. If it isn't—if one wall is contacted ahead of where the cutter stylus was at a given instant and the other wall is contacted behind of where the cutter stylus was at that same instant—there will be a falsification of the information that was cut into the groove. This is commonly called lateral tracking error and the most serious consequence traditionally attributed to it is second harmonic distortion. We, too, accepted that at the time we wrote our original article. But it just isn't so. The most important distortion that results is *time distortion.* The playback stylus is ahead of real-time reproduction on one wall and behind on the other wall. And the ear is considerably more sensitive to that kind of time smear than to harmonic distortion.

Now think about the vertical motion of the cutter stylus. For practical reasons that don't really concern us here, this motion isn't perpendicular to the plane of the disc but at an angle to it, the line of motion leaning in the direction of the groove already cut. Now any up-and-down motion at angle also has a back-and-forth component. On the down stroke, the cutter stylus is adding a tiny amount to the linear speed with which the groove is cut; on the up stroke it subtracts a tiny amount. Aha, you say, time distortion is being cut permanently into the groove. Exactly. But what if the playback stylus is set to move up and down at exactly the same angle as the cutter stylus? Then its own velocity modulation will be synchronized exactly with that of the original cut, and the signal will be reproduced in real time. Again, we must repeat that the ear is enormously sensitive to the time domain (see also our loudspeaker survey in this issue) and that you don't easily get away with any kind of fudging there.

So there's your picture: the playback stylus is riding in the groove, mimicking the motion of the cutter stylus exactly and in all dimensions. That means (1) it must be firmly seated without any loss of contact whatsoever; (2) its points of contact with the left and right wall must both be on the same radius of the disc; and (3) its vertical motion must be at exactly the same angle from the perpendicular as was the cutter's. You can ensure (1) by setting the vertical tracking force as high as possible, as long as the stylus cantilever doesn't collapse. (Contrary to untutored opinion, this will not shorten the life of the record as long as the stylus tip is in good shape and the cartridge design isn't the kind that permits ultrasonic hammer blows by the stylus on the groove wall.) As for (2), that's what lateral tracking alignment is all about, and in a moment we'll give you the most practical and effective alignment instructions available anywhere. When it comes to (3), you're pretty much on your own, but it isn't a completely hopeless situation, as we shall see.

First, however, we must point out how useful this mental image of the correct three-dimensional stylus/groove interface can be in explaining and understanding design requirements for cartridges, arms and turntables. For example, why is a biradial stylus, such as the Shibata, superior to a conical stylus? Obviously because the Shibata's longer and narrower areas of contact with the groove mimic much more closely the geometry of the cutter stylus and permit a finer resolution of the time-related details inherent in the cut. But then why does the conical stylus have so many partisans who claim it sounds better than the Shibata? Because it *does* sound better when the lateral and vertical alignments haven't been properly attended to. The Shibata resolves the consequent deviations from time coherency much more precisely and audibly, whereas the conical stylus glosses over them to some extent. A properly aligned Shibata, on the other hand, eats a properly aligned conical for breakfast when it comes to resolution of detail.

Or take antiskating compensation. Why is it so important? Because the skating force deflects the stylus and creates tracking error (i.e., alters the stylus/groove geometry). The usual explanation that it creates unequal stylus pressure on the two groove walls and therefore unequal indentation is only part of the story. And, for the same reasons, too much antiskating bias is just as bad; it's simply a reversal of skating force in the opposite direction. Now think about tone arms. Why should the arm be absolutely rigid? Because any kind of flexing or axial "pumping" will shift the stylus with respect to the groove. And why should the arm be nonresonant? Because if it can store and release energy in the frequency range relevant to the groove, at least part of the energy will end up driving the stylus in a way that's completely unrelated to the stylus/groove geometry. What about turntable rumble or acoustic feedback? Same thing—the stylus will be dislocated with respect to the groove with some kind of low-frequency periodicity. On the other hand, why is pivot bearing friction a less important consideration? Because as long as it isn't large enough, relative to the vertical tracking force and the stylus compliance, to apply a bias to the stylus and thus alter the stylus/groove geometry, it can't possibly have an effect on the reproduced signal. And so forth and so on: the geometrical (or call it topological) explanation covers everything.

**Can we really hear all this?**

Our recent experiments with fanatically careful alignment to optimize both lateral and vertical tracking geometry have produced astonishing results. Going from routinely conscientious mounting of the tone arm and cartridge, by following the manufacturer's instructions unquestioningly, to optimum alignment, in accordance with the procedures outlined below, made in every case a greater difference in audible quality than changing to another good cartridge, preamp or power amp! The sound just opens up and assumes a totally new character—and the listener's mind reels. In fact, we now recommend that you perform these alignments before you change any component in your system, otherwise you'll never know exactly what kind of sound you're giving up—and switching to. Where this leaves the typical listening tests performed in audio stores, manufacturers' sound rooms, and equipment reviewers' homes we'll leave to your imagination. All we can say is that, whenever someone now tells us that he found component A to be superior to component B after extended listening to reference records, the first thing we ask is, "How did you align your tone arm and cartridge?" If the answer is a blank stare or "Huh?" or "Just like it says in the instructions!", we say thank you and change the subject. (And then we thank our lucky stars that our initial insight into the importance of all this came before our preamp survey was too far under way.)

When you begin to think about the dimensions of the microworld in which the stylus and the groove operate, none of the above is the least bit illogical. The linear speed of the groove, at 33 1/3 RPM, varies between 51 cm/sec and 21 cm/sec from the maximum to the minimum radius of the modern 12-inch LP. Let's take a nice frequency like 4 kHz, where the sensitivity of the human ear is at its peak. One full cycle of 4 kHz therefore occupies anywhere between 0.13 and 0.05 mm. (Note that **The Audio Critic** is going metric; this is as good a place to start as any.) Let's take 0.1 mm as a convenient average value. That's 100 microns. Now let's assume that a 5% time modulation of the signal at this frequency (in other words, a time smear of $\pm 12.5$

uS) can blur the focus of a stereo signal to some extent. (This is not an unreasonable figure to assume.) You're then dealing with a 5-micron linear portion of the groove. That's only 50,000 angstroms; you're approaching optical dimensions! When you consider that the stylus overhang specified in the mounting instructions of the Grace G-707 is 15 mm instead of the correct 17.5 mm and that the unmodified offset angle of the arm is off by more than 1° from optimum (and that this happens to be one of the geometrically best-optimized arms on the market!) you begin to realize that we aren't just whistling Dixie here. There can exist a serious nonsynchronism between the recorded and the reproduced signals.

**How to optimize lateral tracking geometry.**

We're publishing a completely new table of lateral tracking alignments here, simpler and more useful than the Universal Design Graph in our first issue. Not that the latter was in any way incorrect; in numerical data it differs from our new table by less than the least possible measurement error. The main difference is that the maximum and minimum radii of the recorded area assumed in our table are the official IEC values, standard on all Neumann and similar mastering lathes. The graph, which predates the stereo era by several years, was based on very slightly different limiting radii; furthermore, it didn't contemplate the possibility of twisting the cartridge in the headshell, so that it also provides correctly matched overhang values for other than optimum offset angles, an audible compromise that turns out to be quite unnecessary.

***Editor's Note:* We've been informed by Mr. Warren B. Syer, the publisher of *High Fidelity*, that our Universal Design Graph was a reproduction of page 23 of the August 1957 issue of the now defunct *Audiocraft* magazine, with copyright still owned by *High Fidelity*. Needless to say, we committed this infringement unwittingly; a photoprint of the graph had been in our files for all these years, with only handwritten notes to relate it to the original Seagrave equations. Our apologies to *High Fidelity*; our acknowledgment to Dr. John D. Seagrave, the author, and to *Audiocraft* (may it rest in peace).**

There was something else in our first article that needs emendation. We erroneously credited Benjamin B. Bauer and Dr. John D. Seagrave for the original work on the mathematical analysis of tracking error. Actually, their articles added nothing substantially new to the considerably more detailed and profound study of the subject by H. G. Baerwald, an obviously brilliant man who worked for The Brush Development Company in Cleveland, Ohio, and published his massive paper on tracking error and optimal pickup design in the December 1941 issue of the *Journal of the Society of Motion Picture Engineers*, years ahead of anyone else. This is the classic work and the most complete solution of the mathematical problems posed by the subject; the computer program that resulted in our new alignment table is based on the Baerwald equations.

When you look at the table, please note that as the arm length is changed, the optimum overhang changes very, very gradually, whereas the optimum offset angle changes much more rapidly. This is the most important thing to understand in order to develop a good feel for the alignment procedure; it explains, among other things, why you can't just move the cartridge straight back and forth to trim in the overhang, no matter how small an adjustment you're making.

Let's track through the essential steps in using the table.

First, install the cartridge in the arm, but don't tighten the screws all the way. Allow the cartridge to be just barely movable when pushed. For openers, you might as well locate the stylus tip approximately where the instructions tell you to.

Now measure the effective arm length, from lateral swing axis to stylus tip, with an accurate ruler, such a machinist's scale. You don't have to agonize over this measurement, but try to get it right within 0.5 mm (0.02 in.).

Look up the optimum overhang corresponding to this arm length in the table. You may interpolate without any fear of inaccuracy. (For example, if the effective arm length is 238.5 mm, take the halfway point between the 17.398 and 17.319 mm overhangs, i.e., 17.3585 mm, which you'll never measure more accurately than 17 1/3 mm.)

Set this overhang (measured from the stylus tip to the *center* of the turntable spindle

# Table of Optimum Overhang and Offset Angle Alignments for Pivoted Tone Arms

Optimized for a 12-inch LP record with a recorded area between the IEC Standard maximum and minimum radii of 146.050 mm (5.750 in) and 60.325 mm (2.375 in). Zero tracking error in all cases at radii of 120.90 mm (4.76 in) and 66.04 mm (2.60 in). For in-between values of arm length, interpolation error will be smaller than the least possible measurement error.

| Effective Arm Length | | Optimum Overhang | | Optimum Offset Angle | Effective Arm Length | | Optimum Overhang | | Optimum Offset Angle |
|---|---|---|---|---|---|---|---|---|---|
| mm | (in) | mm | (in) | ° | mm | (in) | mm | (in) | ° |
| 200 | (7.874) | 21.055 | (0.829) | 27.854 | 238 | (9.370) | 17.398 | (0.685) | 23.118 |
| 201 | (7.913) | 20.938 | (0.824) | 27.704 | 239 | (9.409) | 17.319 | (0.682) | 23.061 |
| 202 | (7.953) | 20.822 | (0.820) | 27.555 | 240 | (9.449) | 17.241 | (0.679) | 22.914 |
| 203 | (7.992) | 20.708 | (0.815) | 27.408 | 241 | (9.488) | 17.164 | (0.676) | 22.814 |
| 204 | (8.031) | 20.595 | (0.811) | 27.262 | 242 | (9.528) | 17.088 | (0.673) | 22.714 |
| 205 | (8.071) | 20.483 | (0.806) | 27.118 | 243 | (9.567) | 17.012 | (0.670) | 22.616 |
| 206 | (8.110) | 20.373 | (0.802) | 26.976 | 244 | (9.606) | 16.937 | (0.667) | 22.518 |
| 207 | (8.150) | 20.264 | (0.798) | 26.835 | 245 | (9.646) | 16.863 | (0.664) | 22.421 |
| 208 | (8.189) | 20.156 | (0.794) | 26.696 | 246 | (9.685) | 16.790 | (0.661) | 22.325 |
| 209 | (8.228) | 20.049 | (0.789) | 26.558 | 247 | (9.724) | 16.717 | (0.658) | 22.230 |
| 210 | (8.268) | 19.944 | (0.785) | 26.422 | 248 | (9.764) | 16.644 | (0.655) | 22.135 |
| 211 | (8.307) | 19.839 | (0.781) | 26.287 | 249 | (9.803) | 16.573 | (0.652) | 22.042 |
| 212 | (8.346) | 19.736 | (0.777) | 26.153 | 250 | (9.843) | 16.502 | (0.650) | 21.949 |
| 213 | (8.386) | 19.634 | (0.773) | 26.021 | 251 | (9.882) | 16.431 | (0.647) | 21.857 |
| 214 | (8.425) | 19.533 | (0.769) | 25.891 | 252 | (9.921) | 16.362 | (0.644) | 21.766 |
| 215 | (8.465) | 19.433 | (0.765) | 25.762 | 253 | (9.961) | 16.293 | (0.641) | 21.675 |
| 216 | (8.504) | 19.332 | (0.761) | 25.634 | 254 | (10.000) | 16.224 | (0.639) | 21.586 |
| 217 | (8.543) | 19.237 | (0.757) | 25.507 | 255 | (10.039) | 16.156 | (0.636) | 21.497 |
| 218 | (8.583) | 19.140 | (0.754) | 25.382 | 256 | (10.079) | 16.089 | (0.633) | 21.409 |
| 219 | (8.622) | 19.044 | (0.750) | 25.258 | 257 | (10.118) | 16.022 | (0.631) | 21.321 |
| 220 | (8.661) | 18.949 | (0.746) | 25.135 | 258 | (10.157) | 15.956 | (0.628) | 21.235 |
| 221 | (8.701) | 18.856 | (0.742) | 25.013 | 259 | (10.197) | 15.890 | (0.626) | 21.149 |
| 222 | (8.740) | 18.763 | (0.739) | 24.893 | 260 | (10.236) | 15.825 | (0.623) | 21.064 |
| 223 | (8.780) | 18.671 | (0.735) | 24.774 | 261 | (10.276) | 15.761 | (0.621) | 20.979 |
| 224 | (8.819) | 18.580 | (0.731) | 24.656 | 262 | (10.315) | 15.697 | (0.618) | 20.895 |
| 225 | (8.858) | 18.490 | (0.728) | 24.539 | 263 | (10.354) | 15.633 | (0.615) | 20.812 |
| 226 | (8.898) | 18.401 | (0.724) | 24.423 | 264 | (10.394) | 15.570 | (0.613) | 20.730 |
| 227 | (8.937) | 18.313 | (0.721) | 24.309 | 265 | (10.433) | 15.508 | (0.611) | 20.648 |
| 228 | (8.976) | 18.225 | (0.718) | 24.195 | 266 | (10.472) | 15.446 | (0.608) | 20.567 |
| 229 | (9.016) | 18.139 | (0.714) | 24.083 | 267 | (10.512) | 15.384 | (0.606) | 20.486 |
| 230 | (9.055) | 18.053 | (0.711) | 23.971 | 268 | (10.551) | 15.324 | (0.603) | 20.406 |
| 231 | (9.094) | 17.969 | (0.707) | 23.861 | 269 | (10.591) | 15.263 | (0.601) | 20.327 |
| 232 | (9.134) | 17.885 | (0.704) | 23.752 | 270 | (10.630) | 15.203 | (0.599) | 20.248 |
| 233 | (9.173) | 17.801 | (0.701) | 23.644 | 271 | (10.669) | 15.144 | (0.596) | 20.170 |
| 234 | (9.213) | 17.719 | (0.698) | 23.537 | 272 | (10.709) | 15.085 | (0.594) | 20.093 |
| 235 | (9.252) | 17.638 | (0.694) | 23.431 | 273 | (10.748) | 15.026 | (0.592) | 20.016 |
| 236 | (9.291) | 17.557 | (0.691) | 23.325 | 274 | (10.787) | 14.968 | (0.589) | 19.940 |
| 237 | (9.331) | 17.477 | (0.688) | 23.221 | 275 | (10.827) | 14.911 | (0.587) | 19.865 |

and coinciding with the effective arm length) either by moving the cartridge in the headshell (in which case you'll change the arm length slightly) or by moving the entire arm. The overhang can be measured with a short and narrow machinist's scale or ruler; a good way to do it is by measuring up to the outside of the spindle only and then adding the spindle radius, which is half of the standard spindle diameter of 7.163 to 7.214 mm (0.282 to 0.284 in). Thus, in the example above, you'd set the stylus tip 13¾ mm from the *outside* of the spindle.

Now comes the crucial part. Prepare a piece of cardboard (a file card of the right size will do) by punching a spindle hole in it and marking off radial distances of 66.04 mm (2.60 in) and 120.90 mm (4.76 in). You could also do this with one of those convenient (and incorrectly marked) paper protractors furnished by tone arm manufacturers. Then twist the cartridge in the headshell so that you can see absolutely no tracking error at those two points. Keep playing with the cartridge until you get this right. The offset angle will then be automatically correct for your overhang.

After that, set the vertical tracking force to the highest value allowed in the cartridge specs and adjust the antiskating bias. The easiest and most accurate way to do this is by ignoring the bias calibrations on the arm (except as a general guideline) and setting the bias by observing the stylus with a small magnifier or loupe. When tracking a low-level passage, there should be absolutely no difference in the centered (neutral) position of the stylus on or off the record. If you see the tiniest snap as you lift the cartridge off the record, the bias is incorrectly set.

And now comes the most annoying part. After you've done all this, everything is probably slightly off, since all these alignments interact. The antiskating adjustment could quite conceivably have misaligned the two zero-error points, in which case you'll have to trim those in again. Then the overhang will probably be off, and the effective arm length in turn. Keep going around in a circle, from one to the other, until everything is right on the button and in accordance with the table as well as the alignment card. The end result will be worth the trouble.

Please note that these alignments optimize your record player for 12-inch LP records only. Mr. R. R. Mills, the managing director of J. H. Reproducers in Australia, took us to task for not recommending in our first article a dual-purpose optimization for both LP's and 45-RPM doughnut singles. We think that's a preposterous idea: spending in the upper hundreds of dollars for a fine cartridge/arm/turntable combination and then getting more distortion than necessary on your best LP's just so that the bubble-gum cheapies can be played with a little less distortion. Pure LP optimization is barely good enough in our opinion; optimizing for the much smaller minimum radius of the recorded area on 45's carries the pivoted-arm compromise a bit too far for any serious audiophile's comfort.

**What about the vertical tracking angle?**

Here the going gets a bit rough. We find that deviations of as little as 20 minutes of arc from the correct vertical tracking angle are audible, yet there are modern LP records, none of them more than ten years old, cut with angles as small as 15° and as large as 18.5°. Go back a little further in time and the spread becomes 13° to 22°. In the earliest days of stereo there were even a few records cut with a *negative* vertical tracking angle. (You'd have to twist your cartridge around 180° and reverse your left/right leads to play those correctly!) And the angle is never, never labeled on the record. What do you do?

We suggest that you align your cartridge/arm vertically to make the latest state-of-the-art records cut with the Neumann SX 74 head sound their best. That's probably a 17.5° or 18° cut, and you're unlikely to run into anything higher. The Mark Levinson records are a very good example. After having performed every step of the lateral tracking alignment, put on such a record and play with the arm height adjustment until the sound can't be made any cleaner and more transparent. Generally speaking, if the sound is hard and overbright, you've raised the arm too much; if it's muddy and midrangy, you've lowered it too much. Try to get it just right; it isn't so hard with good associated equipment—the right alignment almost clicks into place. And don't look at the arm tube. Its position is meaningless except as an indication of where you've been and where you're going.

Once your vertical tracking alignment is set for the largest angles currently used in the business, you're ahead of the game, since you can always make the angle smaller by shimming up the record (say, with another record) but you can make it larger only by readjusting the arm height. You can even make a series of calibrated cardboard discs and mark on the album cover how many of them you need for shimming up that particular record. Just don't give up because the whole thing is too much bother. Once you've developed a routine, you'll waste very little time and you'll enjoy your record collection a great deal more. Remember, the difference isn't trivial.

**A word about our tone arm and turntable reviews.**

Phono cartridges are very special active devices and will be getting very special consideration beginning in Part II of this series. Tone arms and turntables, on the other hand, are passive vehicles for the cartridge; their performance is defined by their mechanical excellence as tools or machinery. Once you understand the stylus/groove interface, the mystique of purely subjective testing ("the highs are grainy and the midrange is hooded") begins to appear singularly inappropriate to tone arms and turntables. Not that their ultimate proof of quality isn't in the sound. Of course it is. But you don't have to test a hammer that has a five-inch handle by vigorously hammering nails with it for two days. You say, "This hammer doesn't have a decent handle; give me another one." Since the world of tone arms and turntables is full of hammers with five-inch handles, simple inspection often suffices to make a "first cut" in a test program whose sole concern is finding the best. In some cases a few minutes of use will reveal all the necessary information. In other cases it may take weeks or months.

We've taken this completely pragmatic, common-sense approach to evaluating tone arms and turntables. It's entirely possible that in the end a few supersubtle finalists will require careful A-B listening comparisons under controlled conditions. We're far from that point yet. We're still looking for equipment that, right up front, doesn't seem to threaten the stylus/groove relationship in any way whatsoever.

Nor do we want to make any broad generalizations, at this juncture, about design criteria for the ideal tone arm or turntable, other than the integrity of the stylus/groove geometry itself. Under each particular model, we're commenting on whatever specific design problem that unit poses; when the time comes, after we've reviewed a sufficient number of different designs, we'll summarize the lessons to be learned.

Before we get into the reviews, however, we'd like to bring up one amusing example, which will serve to illustrate some of the technical points we're trying to make. Somebody we know made a truly superb tone arm some time ago for a special laboratory project. It consisted of a plastic soda straw, pivoted on four ordinary household pins in a gimbals configuration. Consider this ultrasophisticated device: it had very low mass, great axial rigidity, low Q because of the relative deadness of the plasticized material, only the correct two degrees of freedom in the gimbals (not able to twist every which way like a unipivot), quite adequately low bearing friction at the four points of suspension, and of course no silicone gunk or other damping at the pivot to modulate the vertical tracking force on warps. Of course, the experimental cartridge it was made for was extremely low in mass and high in compliance; for your normal cartridge the straw could be replaced with, say, a straight section of balsa wood or some other heavier material that's rigid and dead. How about that, Dynavector?

Oh yes, just one more thing. The Duxseal we keep referring to in the reviews is the acoustically deadest material known to us, originally made as a sealant for plumbers, electricians, appliance installers, etc., by Johns-Manville. You can buy it from suppliers to that trade. It will damp some of the most stubborn resonances.

# Dual CS721

*United Audio, 120 South Columbus Ave., Mount Vernon, NY 10553. Dual CS721 automatic single-play turntable, $400. One-year warranty; distributor pays return freight. Tested #052570, on loan from distributor.*

This is Dual's top-of-the-line turntable/ arm model, with electronically controlled direct-drive DC motor. We don't find anything seriously wrong with it, except that the combination of the Kenwood KD-500 turntable and Grace G-707 tone arm is even better and costs $60 less (but isn't automatic, of course). Otherwise the Dual CS721 would be our current top recommendation as the record player for a medium-priced stereo system.

Dual is particularly proud of the arm, and it isn't a bad arm at all. The bearings are deployed in an ideal gimbals configuration and appear to be correctly preloaded. We could detect no play or wobble whatsoever. The arm tube is straight, rigid and quite dead. The offset angle of the nonremovable headshell is *almost* correct, requiring minimal twisting of the cartridge for fully optimized alignment. With those advantages, you're already way ahead of the game. We also like the continuously adjustable arm height (i.e., vertical tracking angle) feature, which makes this all-important alignment truly painless by means of a simple knurled knob control. Now if only the control could be used during play . . . but no such luck.

The two weak spots of the arm, in our opinion, are the headshell and the counterweight. The mechanism that clips the removable cartridge holder into the fixed headshell isn't quite positive enough; there remains the possibility of not only play but also a lack of positional repeatability under worst-case conditions (such as frequent removal and reinsertion). You can imagine how much that disturbs a tracking-error fanatic. Nor is the entire headshell assembly dead enough for our taste. As for the counterweight, we question the antiresonant mechanical filters built into it for the purpose of tuning out arm and chassis resonances. The jellylike bobbling of the counterbalance breaks up a resonant peak into several smaller ones (just a few dB lower in amplitude) instead of dissipating it resistively (like carbon fiber, balsa wood or even silicone gunk), which would be definitely more desirable. We aren't suggesting that this is a totally wrongheaded solution, given the design constraints of a very compact, practical, easy-to-use, automatic turntable at a price—but a purist's approach it isn't.

The turntable itself is also of basically good design, without disqualifying faults. We'd like to see a platter that doesn't ring at all when struck; the Dual's isn't quite dead enough. The resonant frequency of the chassis on its suspension could also be a few Hz lower for best results. All in all, though, this is a competent, well thought-out job, and it doesn't appear to suffer from the horrible things direct-drive turntables are sometimes accused of, such as cogging.

The only thing we can't figure out is where the CS721 fits into the audiophile scheme of things, since it's neither the ultimate nor (in view of Kenwood/Grace) the best for the money. How about best for the money if you must have an automatic?

# Dynavector DV-505

*Onlife Research, Inc., Tokyo, Japan. Distributed in the U.S.A. by Audioanalyst, Inc., PO Box 262, Brookfield, CT 06804. Dynavector DV-505 tone arm, $575. Tested #510880, owned by The Audio Critic.*

Since we've already discussed this "ultimate" arm (its name in Japan is Ultimo) at some length in two previous issues, we only want to summarize our conclusions here.

Despite its sci-fi appearance and mind-boggling price tag, it isn't the ultimate arm. It suffers both from errors of design and from errors of execution. The massive, heavily damped main arm, pivoted to move only laterally, and the much lighter, undamped subarm, pivoted to move only vertically, constitute an erroneous design concept in our opinion. Regardless of the convoluted rationale presented in Onlife's technical brochure, such a gross difference between the lateral and vertical motional impedances will tend to resolve vectors with a strong upward bias, thus falsifying the information cut into the groove. Furthermore, the sensitivity of the short subarm to warp wow is inevitably greater than that of any normal arm.

The greatest boo-boo in the physical execution of the DV-505 is the way the arm height adjustment mechanism is slotted and keyed. Every time you change the arm height with this beautifully convenient device, the play between the key and the slot results in a shift of the lateral swing axis of the arm, completely negating the overhang alignment. Once you're aware of this unbelievable defect, you can push the damn thing back to the end point of the play, hoping that at least that point is stable. But for $575? Come on, Onlife.

We think we have figured out the reason why this arm so often appears to improve the sound when inserted into a known system. The unusually large mass of the entire structure tends to act as an anchor or inertial platform, dissipating some of the grosser extraneous disturbances of the stylus/groove relationship that afflict the typical installation. As soon as purposeful and meticulous care is taken to eliminate these disturbances (better mechanical and acoustic feedback isolation of the turntable, for example, or a dab of deading material such as Duxseal in the right places), correctly designed arms such as the Grace G-707 begin to sound better. It's as simple as that.

There are more ways to skin a cat than hanging a $575 anchor around its neck.

# Grace G-707

*Sumiko Incorporated, PO Box 5046, Berkeley, CA 94705. Grace G-707 tone arm, $140. One-year warranty; customer pays all freight. Tested sample owned by The Audio Critic.*

We consider this simple, ungimmicky arm to be virtually unexceptionable in design; refinements would be possible, of course, but mainly in execution rather than concept. It's the best all-around workhorse arm we've been able to find so far. You must keep in mind, however, that it has a fixed headshell, so that frequent cartridge changes are a nuisnce. And, needless to say, you still have to twist the cartridge in the headshell every time, since the lateral geometry still isn't right on the money.

Interestingly enough, this is also a straight-tube, gimbals-suspension design, like the Dual arm, and there's really no better approach. The Grace is a little longer and a little simpler, both of which are fundamental advantages; its headshell and arm tube seem to be reasonably dead, although two little dabs of Duxseal (one between the cartridge screws on top and one on the arm fairly close to the pivot) resulted in an ever-so-slight increase in sonic clarity, proving that there was room for improvement even in that department. The slight extra mass also brought the G-707 right in line with the compliance of our Denon DL-103S cartridge; overall, however, the arm appears to be acceptably compatible in mass with a wide range of modern cartridges and is certainly not too heavy for any that we know of. As for the bearings, they appear to be eminently free from play and wobble, which probably accounts for a significant part of the arm's audible performance.

We could wish for generally more rugged construction, a more convenient antiskating adjustment, a cueing mechanism with more positive feel, etc., etc.; but the fact remains that this is the arm we're currently using in our reference system, even though we can't help wondering why there isn't this kind of intelligent design available along with SAEC-like construction—for those who are willing to pay the price.

# Grace G-940

*Sumiko Incorporated, PO Box 5046, Berkeley, CA 94705. Grace G-940 tone arm, $150. One-year warranty; customer pays all freight. Tested sample owned by The Audio Critic.*

This isn't exactly our cup of tea: a unipivot design (i.e., free to move in various undesirable ways that would be impossible with a gimbals-type suspension); fluid-damped at the pivot, which is far from the ideal place; and without antiskating bias, so that there will be lots of tracking error (among other things), no matter how carefully the overhang and offset angle are aligned. What's more, the arm is on the heavy side for some of the latest high-compliance cartridges.

The fact that the G-940 still sounds a lot better than most other arms tells a great deal about most other arms.

# Harman Kardon Rabco ST-7

*Harman Kardon, 55 Ames Court, Plainview, NY 11803. Rabco ST-7 straight-line tracking turntable, $430. Tested #2423689 and #3000927, owned by The Audio Critic.*

If this were a successfully executed product, more than half of our preceding article, including the overhang and offset-angle table, would be unnecessary or at least academic. The ST-7 has a straight-line tracking arm, and that means no overhang and no offset (and therefore no skating force, either). The ideal lateral geometry. And it's a fully automated single-play turntable, with universal arm and two-speed table lovingly made for each other—obviously the ultimate fruition of the original Rabco idea, right? Wrong.

Our tests of two different samples were nipped in the bud by an absolutely intolerable and disqualifying defect common to both, which we must therefore assume to be a design characteristic. The carriage of the arm transport mechanism is so loose and wobbly that the theoretically ideal stylus/groove relationship made possible by straight-line tracking is totally negated, indeed made worse than with just about any conventional arm. The result is a big resonance in the upper-bass/lower-midrange region, adding an insistent coloration to the sound that we don't find at all acceptable in advanced audio equipment.

The overall construction of the ST-7 seemed much too flimsy to us from the beginning, but this we didn't expect. Next!

# Infinity 'Black Widow'

*Infinity Systems, Inc., 7930 Deering Avenue, Canoga Park, CA 91304. 'Black Widow' tone arm, $200. One-year warranty; customer pays all freight. Tested #11216, on loan from manufacturer.*

Aside from its Bongiornoesque name, we found this slender and not particularly lethal-looking tone arm quite livable with on first acquaintance, even though twisting the cartridge for optimum geometry in the tiny (and, of course, not quite correctly offset) vestige of a headshell took some doing. Then we discovered that damping this half-shell as well as the bearing housing with some Duxseal resulted in a substantial improvement in transparency and definition, so it's an inescapable conclusion that the arm rings to some degree. Closer examination of the bearings indicates some proneness to jitter; what's more, the little half-headshell can be plucked to twang ever so slightly; both of which resonant conditions may have been helped by the damping.

Overall, the arm seems even a little flimsier than the Grace G-707, which for 30% less offers substantially the same virtues without the same vices.

# Kenwood KD-500

*Kenwood, 15777 South Broadway, Gardena, CA 90248. KD-500 direct-drive turntable, $199.95. One-year warranty; customer pays all freight. Tested #430148, on loan from owner.*

Although we haven't tested some of the top contenders yet (for example, the Linn Sondek LP12 and the Thorens TD-126 Mk II are among those next in line), we doubt if we'll ever find a better *value* among turntables than this excellent and remarkably low-priced Kenwood unit. There really isn't all that much to say about it, except that it's entirely trouble-free and makes good cartridges and tone arms sound their best. For example, we like the sound of our Denon DL-103S/Grace G-707 combination much better on the Kenwood than on the Luxman, which costs three times as much.

The KD-500 is living proof that what makes a turntable sound good isn't exotic engineering, super specs or fancy features, but simply a design that allows the stylus/groove relationship to remain undisturbed at all times. The "resin concrete" chassis on which all parts are mounted is probably a strong contributing factor; it seems to be nice and dead. The platter itself is rather live (it rings like a bell when struck) and could probably benefit from a liberally applied layer of Duxseal underneath. (Unfortunately, we don't own ours, so we can't mess with it.) The adjustable legs of the Kenwood aren't the most sophisticated kind either, and we had to place ours on a marble slab to improve the mechanical feedback isolation. Furthermore, the direct-drive DC motor doesn't have much torque. But who cares? Properly set up, the Kenwood seems to be more stable and feed less crud to the cartridge and arm than any other turntable we've tried so far.

As one of the old perfume ads used to say, it's unfair but it works.

# Luxman PD-121

*Lux Audio of America, Ltd., 200 Aerial Way, Syosset, NY 11791. Model PD-121 direct-drive turntable, $595. Three-year warranty; manufacturer pays all freight. Tested #E6901267 and #F6901268, owned by The Audio Critic.*

This was our reference turntable for many months, mainly because nothing more precise and beautiful had ever been built, and it worked very reliably. All right, the mechanical feedback isolation wasn't the greatest, but a marble slab took care of that. (What do you think marble was invented for? Michelangelo?)

Deep down we always knew, however, that there was something subtly wrong with the sound. A kind of thickness in the upper bass and lower midrange. We tried to blame everything—record, cartridge, arm, etc.—except this gorgeous machine. But the day of reckoning finally came. The unspeakable truth: the PD-121 drums. Like a real live drum. That large, flat base just isn't dead enough. The main cause of excitation is airborne acoustic feedback, but you can verify the character of the resonance by tapping on the turntable just so, with the record playing and the volume control up. The coloration of the resulting undamped transient is the same as the coloration we heard in the music. And the drumming isn't curable by any simple means. Goodbye, PD-121.

(And you thought it would be something really sophisticated, like cogging or some other fashionable direct-drive vice, didn't you?)

# Mayware Formula 4

*Mayware, England, distributed in the U.S.A. by Polk Audio, 1205 South Carey Street, Baltimore, MD 21230. Formula 4 PLS4/D tone arm, $149.95. Tested sample owned by The Audio Critic.*

This is a unipivot arm, with silicone damping applied right at the pivot (i.e., in series with the stylus compliance), and you know from our Grace G-940 review above how we feel about both of those design approaches. The Formula 4 also has a somewhat wobbly sliding weight between the pivot and the headshell, so that not only the vertical tracking force but also the dynamic mass of the arm can be varied. An interesting idea, but we aren't convinced that the slight play between this cursor weight and the arm tube isn't audible. Nor that the very light, fixed headshell doesn't ring. Nor that the ultrathin arm tube is rigid enough. Nor that eccentric counterweight doesn't set up some torsional forces that aren't completely balanced out.

Since the Grace G-707 is a simpler and more effective design, as well as quite compatible with the same high-compliance cartridges that the low-mass Formula 4 was designed for, there's really no need to belabor the above points. The Formula 4 also costs a few dollars more and is even less rugged in construction.

J. H. Reproducers Co., an Australian manufacturer, claims that the Mayware Formula 4 is an inferior rip-off of their original design. We're reporting this merely as a partisan allegation, since we haven't been able to get our hands on the J. H. version thus far to form our own opinion. Needless to say, however—once a fluid-damped unipivot, always a fluid-damped unipivot.

# SAEC WE-308 NEW

*Audio Engineering Corp., Tokyo, Japan. Distributed in the U.S.A. by C. M. International, 4131 Calle de Primera, Torrance, CA 90505. SAEC WE-308 NEW double-knife-edge tone arm, $195. Tested #FJT4955, owned by The Audio Critic.*

This incredibly frustrating combination of precision workmanship and geometrical blundering, which we can only liken to a superb Swiss timepiece with an 11-hour dial, was already discussed in sufficient detail in our CES report (May/June 1977, page 17) to constitute a review; we really don't feel like beating up on these obviously dedicated though misguided people all over again. We must add, however, that we finally succeeded in twisting the Denon DL-103S cartridge into the correctly aligned position on the SAEC; if the offset angle of the arm had been just half a degree smaller yet, it would have been impossible—and probably *is* impossible with certain other cartridge configurations.

And what do you know, the correctly aligned SAEC/Denon combination still didn't work properly. The mass of the arm was much too high; the subsonic resonance went way down to the point where it coincided with the slight warp excitations produced by the record, and the stylus bobbled like very slow jelly in the groove. Now the Denon isn't a particularly high-compliance cartridge; it can live happily with some fairly massive arms. So it seems that the SAEC is designed only for the very lowest compliances to be found today. Of course who says it was designed at all? (Sorry. There we go again.)

One nice feature of the SAEC, though, is the optional arm stabilizer accessory (AS-500), a large cylindrical weight that threads right onto the arm post under the turntable. It's again the anchor or inertial platform concept, but a lot cheaper and more convenient than in the case of the Dynavector.

Let's be thankful for a good idea even if we can't be for a good arm.

# Recommendations

It's much too early in the game to categorize any tone arm or turntable as the best, even with qualifications. We just haven't looked into enough of them. But at least two excellent and relatively low-priced pieces of equipment emerge from our investigation so far. To those who are unwilling to wait for our reports in the next few issues, we can recommend them unhesitatingly.

**Tone arm: Grace G-707.**

**Turntable: Kenwood KD-500.**

*Editor's Note: Max Wilcox has been busy with an unusually crowded schedule of recording dates. His third article on why records sound the way they do has therefore been postponed until the next issue. Meanwhile, we present our first record review column.*

# The Mark Levinson Recordings

Not too many audio enthusiasts and music lovers are aware of Mark Levinson Acoustic Recordings Ltd., a company operated by the same music addict and compulsive audio perfectionist as Mark Levinson Audio Systems Ltd. Even in stores where you have no trouble finding copies of Sheffield, Audio Lab, Ark and other audiophile-oriented labels, the understated gray jackets of MLAR are generally not visible. This low profile is unlikely to persist, however, since MLAR records are even more clearly superior to others in their sonic qualities than MLAS audio components. The latter have occasional competition; the former have very little. You could call them state-of-the-art. And sooner or later you'll see them on all the more expensive turntables.

At the moment, these records are stocked only by Mark Levinson audio dealers, of which there are fewer than twenty from coast to coast. Broader distribution is being organized; meanwhile, if you're unable to find them, we suggest you write directly to the company at 55 Circular Avenue, Hamden, Connecticut 06514. They'll make sure you'll get what you want. The price isn't cheap: $15 per record, so that for example the "Art of the Fugue" album, consisting of four discs (eight sides), will set you back $60. Mark never believed in giving it away.

But the records are beautifully made. The recording equipment in each case consisted of Bruel & Kjaer 4133 condenser microphones (peak-free out to 40 kHz—we use one for measuring speakers with) and a specially modified Studer A80 tape deck recording on 1-inch tape (¼-inch tracks) at 30 IPS. No Dolby, dbx or any of that jazz. No console. Just straight in. The resulting distortion levels, signal-to-noise ratio and headroom exceed anything currently attainable in disc cutting and pressing, so that the rigors of direct-to-disc recording become an irrelevance. It's the best of both worlds, permitting tape editing as well as live-quality cutting.

Only two channels are used for the stereo information, so there's no possibility of confused imaging as long as the mikes are set up right (which they generally are). Two more channels may or may not be mixed in for ambience information only. That's all. Never more than four mikes and four channels. The mixing takes place directly into the cutter amplifier via an utterly simple mixer box made out of modules similar to those in the Levinson preamps. The cutter head is the incomparable Neumann SX 74; the cutting is performed by the incomparable Bob Ludwig. The discs are pressed in France by the most careful outfit Mark could find for the purpose. (We heard he was almost thrown out of there because he pushed and nitpicked even these dedicated people a bit more than they liked. But he got what he wanted.)

So there you are. Absolutely quiet records, tremendous dynamic range, no distortion to speak of, no peaky mikes, no multimiked jumble. Just the perfectly natural sound of some highly competent musicians. If these same musicians were *great* instead of highly competent—if the organist were Walcha, the pianist Horowitz, and so forth—the experience would be simply overwhelming. As it is, it's merely in the not-to-be-missed category. (Why can't Mark Levinson be the VP in charge of production at Columbia or Deutsche Grammophon?)

**Volume One**

*J.S. Bach: The Six Schubler Chorales (Myrtle Regier, organ); Prelude in E-flat Major (Britt Wheeler, organ); three choir selections (Battell Chapel Choir, Charles Krigbaum, director).*

This is the only Mark Levinson recording not made on the Studer A80, yet it's one of the best. Obviously, the 15-IPS ¼-inch machine used was a good one. The organ is captured in its full glory, with low pedal notes such as you seldom if ever hear from a phonograph record. It's the von Beckerath organ at Yale, incidentally; not the greatest baroque organ in the world, not even von Beckerath's best effort, but beautiful just the same and beautifully recorded in actual performances before an audience, with just the right liveness. The choir selections are also of you-are-there quality; performances are musicianly but not thrilling. Every red-blooded audiophile should own this record.

**Volume Two**

*Maurice Ravel: Valses Nobles et Sentimentales. Joseph Haydn: Sonata No. 49 in E-flat Major. Lois Shapiro, piano.*

Technically this is perhaps the least spectacular of the Mark Levinson recordings, although the piano sound is super clean and the peaks are never clipped. It's just that we'd prefer a more expansive, high-ceilinged ambience, with greater depth and roundness. The music, of course, is wonderful; Ms. Shapiro does an especially stylish job on the sonata, which is a superb piece (late Haydn) and alone worth the price of admission.

**Volume Three**

*Bill Elgart: A Life (percussion piece in ten movements). Performed by the composer.*

We aren't fully sold on this composition as a work of art: two LP sides of drumming and percussing, in a rather tight and dry idiom, often suggesting speech rhythms (we think). Very controlled, deliberate, intellectual; not much exuberance. Edgard Varese and Henry Cowell are more varied and more fun. As a recording, though, it's mind-blowing—absolutely the highest-fidelity percussion record known to us. The drums, cymbals, bells etc., are *there*, right in front of you. Unbelievable transients and dynamics, unbelievably quiet background, no grit or crud whatsoever. Required demo material, even if musically it doesn't turn you on. But maybe it will.

### Volume Four

*Vivaldi, Bach, Hindemith, Debussy, Ives, Lennon/McCartney, et al: music for brass or transcribed for brass. (New Haven Brass Quintet, 2 records.)*

This is one of our favorites: two trumpets, horn, trombone and tuba playing everything but the kitchen sink, from sixteenth-century brass music to "Magical Mystery Tour" and "Penny Lane" as if arranged by Gabrieli! Four sides of gorgeous brass sound, guaranteed to make you miserably unhappy with the midrange of your system, unless it's super special. There's a transcription of a Charles Ives song, "Slugging a Vampire," which is almost impossible to reproduce except with the finest equipment from stylus tip to speaker diaphragm, in which case it sounds absolutely stunning. An acid test. The playing is again expert but not sensational; the music ranges from good fun to very beautiful and back again. Put this on your HQD System and feel superior to the Dahlquist/Ampzilla crowd, who will have trouble with it.

### Volume Five

*J.S. Bach: The Art of the Fugue (Charles Krigbaum, organ; 4 records).*

Bach's last and definitive statement on counterpoint, left in open score without a specified instrument or instruments, is performed here on the same von Beckerath organ at Yale as Volume One in the MLAR series. The organist, Charles Krigbaum, is said to be a highly specialized student of this masterpiece, and his performance is certainly beautifully organized and note-perfect. We find his choice of registration a bit monotonous; he also plays everything invariably legato. Even though a true staccato is mechanically impossible on the organ, the choice of lighter, more flutelike stops, more *detache* phrasing and more distinct ictus in certain passages would have made for livelier listening—in the opinion of this Bach lover who doesn't claim to be a scholar. The recording itself is everything an organ recording should be, weighty and ultratransparent at the same time, with a fine sense of acoustical space. The music is so commanding, though, that this can't be considered a demo record (or records); it's frivolous to talk about audio in the presence of Bach's most profound utterances. Quite an album, and quite a Christmas present for the highbrow audio freak.

—Ed.

**HELP STAMP OUT**
**CHEAP PHILIPS-HEAD SCREWS**
**IN AUDIO EQUIPMENT!**

# The Admonitor

*Editor's Note: We have a stack of letters telling us either that (a) this is one of our best features and we must never stop bushwhacking the snake-oil peddlers or that (b) any audiophile with the brains of a gnat can see through the industry's advertising shenanigans and doesn't need our heavy-handed protection. The reason why we're inclined to continue monitoring and admonishing (and occasionally even commending) in this space is that hi-fi ads are a source of not only specific product hypes but also general information about audio. Many of the simplisms, half-truths and specious dogmas that infest the high-end audio scene are recycled versions of statements originally made in ads. And the unwitting carriers of this infectious misinformation are often the same sophisticated consumers who can't be swayed directly by slick advertising and proudly proclaim their skepticism.*

**Dual**

In our first *Admonitor* column, we commended Dual for their informative, factual advertising of the advantages of straight tone arms (with offset headshells) over curved arms. If this had anything to do with their decision to launch their current ads about "the world's best-designed tonearm," we're truly sorry. That's the Marantz kind of language (recent Superscope era, that is). As you can see from our Dual CS721 turntable review elsewhere in this issue, the Dual arm, good as it is, isn't the world's best. Nor, apparently, is Dual's advertising judgment.

**Epicure 20+**

"Before you spend $600 on a pair of speakers, spend $15 on a ticket to Carnegie Hall," says the headline of Epicure's current ad for their Model 20+ speaker system. The ad goes on to explain, in words and in pictures, that in Carnegie Hall or any other concert hall you hear a "first-arrival" sound directly from the orchestra on the stage and then a "second-arrival" sound from the ceiling and other reflective surfaces of the hall. And, wouldn't you know it, the Epicure 20+ faithfully duplicates this real-life sound field: its front-firing woofer/tweeter module gives you the first-arrival sound and its angled, top-firing woofer/tweeter module the second-arrival sound. Just like Carnegie Hall, see?

This seems to be the latest incarnation of The Great Bose Non Sequitur—the argument that, since live music is a combination of direct and reflected sound, an accurate transducer must therefore be both a direct and a reflected radiation source. The absurdity of this line of reasoning becomes instantly apparent when you consider that it makes the finest electrostatic headphones "inaccurate." All direct radiation into your ear, nothing reflected—can't have any real fidelity, right?

The fact is that a direct/reflecting speaker system such as the Epicure 20+ *has* to create a time smear, since the deliberately reflected components of its output reach the ear a split second later than the direct components. It's obvious that, on a record or tape, the sound reflected off the ceiling and the walls of the concert hall is *already recorded in real time* with respect to the direct sound of the orchestra. To hear it accurately, as captured by the microphones, you don't want it further dispersed in time, i.e., reflected all over again. When the ad says that "the Epicure 20+ successfully integrates first-arrival and second-arrival sound," it really means that the speaker is stretching out an event longer than it took in real life. In other words, the sound is blurred.

We could say, "But what did you expect from the makers of the Model Four preamplifier?"—but that would be rubbing it in, so we won't say it.

**Sony SSU-3000 and SSU-4000**

Can a mass-producer of TV sets and transistor radios find happiness in the advanced audiophile market? With their $1000 V-FET power amp and $900 turntable, Sony certainly seems to think so, but one of their recent speaker ads really blew it.

"After people learn what we've done, no one will heckle our speakers," proclaims the headline of a double-page spread advertising the $600-a-pair Sony SSU-3000 and the $800-a-pair SSU-4000. Obviously the discovery that the word "speaker" can have a double meaning was both new and irresistibly delightful to the ad writer, who then maintains the same level of authority and credibility throughout the ad. The part that really got to us, though, goes as follows:

"*You can hear how pure water is.* The purity of the water in which the pulp for the speaker cone is pressed will influence the sound. (Spring water is the best.)

"But water purity would hardly change the frequency response—or any other measurable characteristic.

"Nor would the dye used to color the cone —or the glue used in gluing the cabinet.

"But you'd hear the dye and the glue."

And a little later: "We built a plant . . . at a place called Kofu. Which is at the base of Mt. Fuji. Where we can get all the spring water we want."

Now really. How could a highly professional outfit like Sony allow something like this to get into print? It's precisely the made-by-elves-in-the-Black-Forest kind of mystification that rots the mind of the untutored, gullible consumer of technology.

Get this straight: A *complete* set of measurements will *completely* specify the physical characteristics of a speaker cone—or a cabinet —and therefore its sound. Complete, of course, means more than just frequency response. If a change in the water or the dye changes, say, the mass of the cone by a small amount, or its stiffness, the change may be audible—but you can measure it if you know what to measure. And it doesn't matter what kind of glue holds the cabinet together (it could be Clark Kent squeezing it with his hands); as long as the joints can't move, the cabinet will sound the same. If they *can* move, there are measurements that will show it. There's no mystery, no kitchen secret, no French chef's fingertip-kissing *je ne sais quoi* to making a cone or building a speaker cabinet. (See also the subwoofer article in our second issue.)

Of course, it's entirely possible that if this ad were traced back from the ad agency through the Sony ad manager to the engineer who provided the original briefing, the latter might protest that he never said these things, at least not the way they got into print. That would be quite typical of the advertising process.

The fact remains, however, that the ad got through, these inanities were published, and the harm is done. Sony—no baloney?

# Classified Advertising

***Rates:** For 25 cents per word, you reach everybody who is crazy enough (about accurate sound reproduction) to subscribe to* The Audio Critic. *Abbreviations, prices, phone numbers, etc., count as one word. Zip codes are free (just to make sure you won't omit yours to save a quarter). Only subscribers may advertise, and no ad for a commercially sold product or service will be accepted.*

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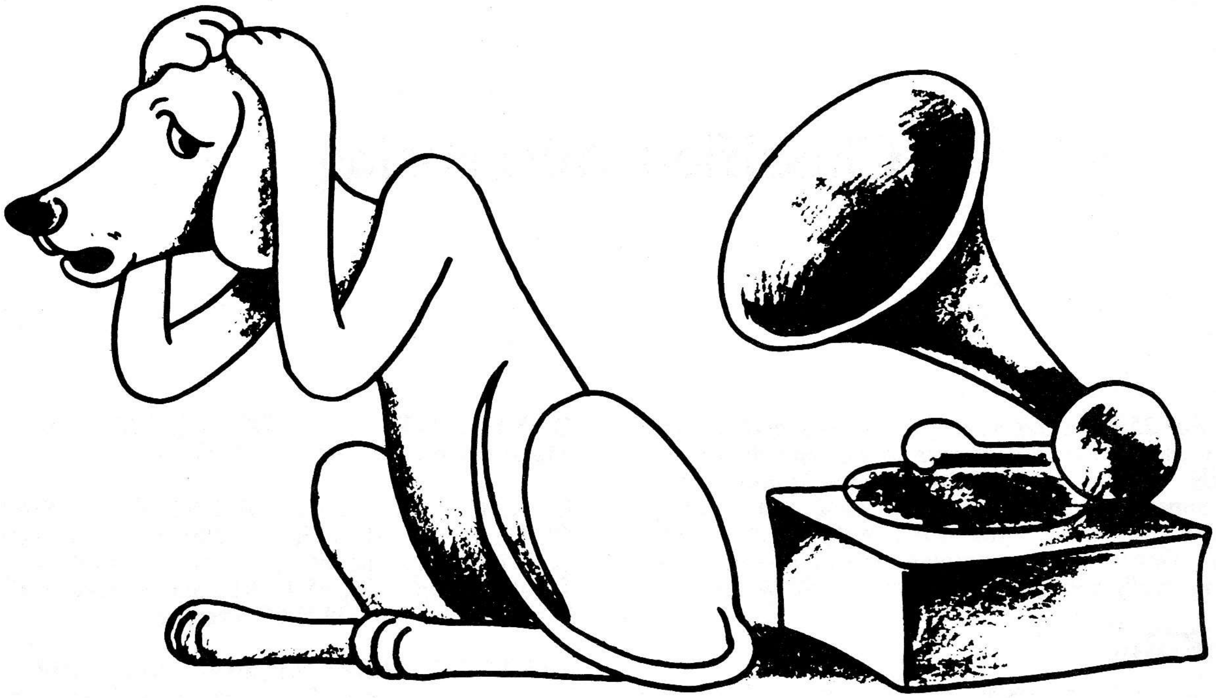
QUINTESSENCE PREAMPLIFIER, Thorens TD-126C, Yamaha HP-1's. For any used component in mint condition, call (401) 521-2072.

DAYTON WRIGHT ELECTROSTATIC XG-8 Mk 3. Make reasonable offer. (412) 683-9550.

DYNACO KITS: Stereo 300, $260; PAT-5, $160; SCA-50. Heath AJ-1515 (wired), $300. Realistic 2000, $300; SCT-11. Only one of each new, warranties included. Jonathan, Apt. 6M, 45-10 Kissena Boulevard, Flushing, NY 11355. (212) 539-8060, evenings.

PARAGON 12 PREAMP, $650. 2 Rogers BBC Monitor (LS3/5A) speakers, $370. Janis W-1 woofer, $475. Denon 103C cartridge with DB Systems head amp, $180. Good stuff, great condition. Chuck Cabell, 6223 Kellogg Drive, McLean, VA 22101. (703) 356-4701.

NEED TUITION MONEY FAST. All equipment new. Mark Levinson JC-2D, $995. Ampzilla II, $785. Marantz 10B, $600. Allison I's, $625. Sondek with Decca arm and Mk VI gold cartridge, Audio Research SP-3A-1 and D76A, $1650. A. Cohn, PO Box 17-203, Bishops Corner, West Hartford, CT 06117. (203) 233-2007.



**The End**

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The Audio Critic

## In the next issue:

The speaker survey continues, the plot thickens, and the end is nowhere in sight.

We begin our investigation of headphones, with special attention to electrostatics.

We bring our power amplifier survey to an end (at least for the moment), firming up the conclusions of our laboratory tests.

A whole new batch of preamps, including some important ones. (Sorry—they keep coming!)

And, for the first time, phono cartridges, along with more arm and turntable reviews.

Not to mention the usual extras.