Volume 2, Number 2 Summer/Fall/Year-End 1979

The Audio Critic®

Retail price: $6

In this issue:

We conclude our already notorious 15-hour seminar on the State of the Art with Part II of the edited transcript. It rocks the boat.

We review a $175 integrated amplifier that sounds a great deal cleaner and more like music than many preamp/power-amp combinations costing $2000 and more.

Speaker wires and audio cables are demythologized and made answerable to the laws of physics.

Our price-no-object Reference A and best-per-dollar Reference B systems are revised and updated.

Plus a record number of new equipment reviews, as well as our usual columns and features.

The Audio Critic

Volume 2, Number 2 Summer/Fall/Year-End 1979

The Audio Critic® is an advisory service and technical review for consumers of sophisticated audio equipment. It is published periodically, at irregular intervals averaging between four and five months, by The Audio Critic, Inc., and is available by subscription only (except for limited sales of individual copies at a higher price in selected audio stores). 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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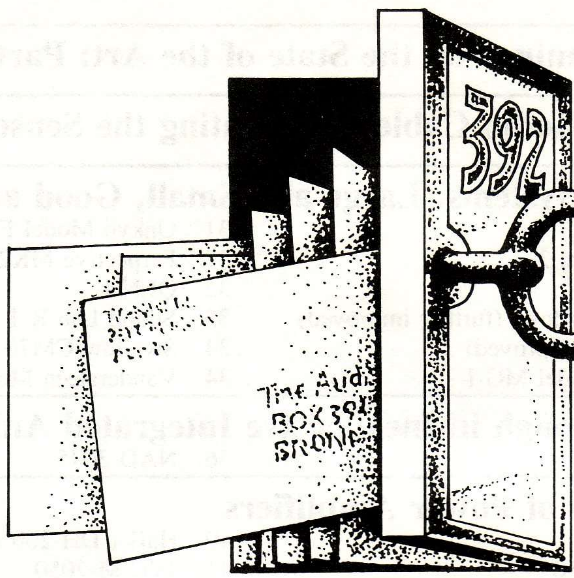
Box 392

Letters to the Editor



Letters from manufacturers in response to our reviews or to other comments about their equipment are usually published unabridged and unedited in this column. So is the correspondence of audio professionals on specific technical, musical or philosophical subjects. Letters of general interest from readers 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.

Again, we open with the letters on what appears to have become the trademark topic of The Audio Critic, phono pickup tracking geometry.

The Audio Critic:

I was amused to read your derogatory reply to my letter in your Winter/Spring 1979 issue. You are quite correct when you state that tracking error at the outer groove is irrelevant, and it is I agree within limits of little importance.

However, I have to advise you that the amount of overhang adjustment available in our headshell means that a cartridge can be aligned for zero error at various distances from the turntable spindle, including your quoted one. We specify 63.500 mm in our setting up instructions, as it happens to be the correct distance for minimum distortion due to lateral tracking error optimised for 12'' discs. Our figure can be found in articles by J.K. Stevenson, *Wireless World*, May and June 1966. To the best of our knowledge Stevenson's work is the definitive one on tonearm geometry, and the articles also refer to Baerwald's much earlier 1941 *JSMPE* work. The difference between your figure and ours is of little importance and is due purely to the variations of the discs. You have chosen to specify the ''representative'' inner design value whilst our ''minimum'' design value obtained from measurements by Stevenson on a batch of records of different makes and different types of music covers all extreme distances in exceptional cases as well, therefore we suggest that our figure is to be preferred.

Finally out offset angle is within the correct range specified by Stevenson.

Sincerely,
Gerald Bearman
Director
Mayware Ltd. (Formula 4)
England

1. Our reply was not derogatory by any standard; it was a simple recitation of facts which you apparently found uncomfortable.

2. What we called irrelevant was not tracking error at the outer groove but your question about it. Tracking error is highly relevant at all points across the record and is predetermined by the laws of geometry.

3. There is no discrepancy among the serious writers on the subject of lateral tracking error (Lofgren, Baerwald, Bauer, Seagrave, Stevenson or anyone else) as to the basic geometrical problem and its proper solution. Thus there is no definitive work on the subject in the sense that one is more correct than any of the others, although Baerwald's paper almost completely preempts the available analyses.

4. As long as the correct mathematical understanding is applied, the only differences that can arise are due, as you yourself suggest, to the chosen maximum and minimum radii of the record between which lateral tracking is to be optimized. What you conveniently forget to mention, however, is that in modern recording practice these two radii are defined by the IEC Standard for 12-inch LP records. The alignment tables published by ***The Audio Critic*** *show the correct figures obtained when this standard is adhered to—as it is by all professional record makers today—whereas your figures are then incorrect.*

5. As we've pointed out once before, messing with the alignment values to obtain a compromise for a broader range of records than just the standard LP's will result in considerably increased distortion on the latter, which after all represent the bulk of any serious audiophile's collection.

—Ed.

The Audio Critic:

Thank you for printing my further letter about the careful use of language in your Winter/Spring 1979 issue. I was rather touched to see you donning Columbo's clothes in order to place a charge of "first-degree intellectual weaseling", but if that delightful bumbling detective had indeed attempted to arrest me on the suspicion that I had not really studied Baerwald, Bauer, Woodward and Cooper, I would have referred him to the Library of Congress—where I trust they keep back copies of *Hi-Fi News*.

In the course of a series of six articles entitled "Pickup Problems", spanning December 1962 to May 1963, I had occasion to quote the Baerwald, Bauer and Madsen formulae in a footnote to Part 2 ('Tracking Error') and included Fox and Woodward's two January 1963 *JAES* articles in the substantial bibliography appended to Part 6 ('Towards the Perfect Pickup'). Of course this doesn't mean that I actually *understood* what they were on about (being a mere engineer-turned-technical-author), but I hope that Columbo would at least hesitate and mumble a bit before making up his mind.

Yours faithfully,
John Crabbe
Editor
Hi-Fi News & Record Review
Croydon, England

Columbo would not hesitate. He would at this point immediately say something like "You know, Mr. Crabbe, I admire you—you've got class." As this column amply testifies, not many audio practitioners are able to respond to technical criticism with this kind of Britannic aplomb.

We would just like to be sure, as we shake hands and part as friends, that the central fact of pickup tracking geometry has not been obscured by the fun and games in these letters: namely that tracking error, either lateral or vertical, creates time-dispersive distortions, which are more irritating than simple harmonic or IM distortion.

—Ed.

* * *

Next, miscellaneous other subjects as seen by the trade.

The Audio Critic:

"The truth, the whole truth and nothing but the truth" is the slogan of The Audio Critic.

Unfortunately, in your efforts to bring to the reader's attention new and exciting developments in the field, you also do them and their dealers an occasional disservice by failing to disclose the whole truth: product availability.

If John Doe develops a new concept and technology in the production of a speaker or amplifier, a JD-1, it is your responsibility to bring this to our attention. Your review tells the audiophile that something better has been made and pushes the other manufacturers to work harder.

But all too often, while the new technology may have just arrived, production is still "around the corner." John Doe cannot make more than two or three JD-1's a month, be it for lack of capital, or an inability to get "mass production" of some unique, previously hand-crafted component.

By your recommendation, the audiophile seeks the best; that is—the JD-1. From all over the world consumers and dealers, relying on your "whole truth," send in deposits, often never to hear from the manufacturer again. Furthermore, dealers, both domestic and foreign, lose business as their customers try to buy the unproduced JD-1. The audiophile is not purchasing what is available and already paid for by the dealer (and probably recommended by you last month).

After all, hi-end audio is as much a business as mid-fi. Both the dealers who have made an investment in the products you praise and the manufacturers who are actually producing these products must be given the opportunity to sell them to the audiophile without competing against products which may never make their way to the market place. We all need sales to stay in business so old products can be produced for your readers to buy and new ones for you to review.

The credibility of the whole hi-end audio industry is at stake; this includes manufacturers, dealers and reviewers alike. If John Doe can make a fantastic new piece of equipment, you do have an obligation to review it. But it must be kept in the proper perspective: product availability as well as product superiority.

Sincerely yours,
Michael T. Berns
President
M. Berns Industries, Inc.
New York, NY

*Right on. But "proper perspective" should also include the realization that **The Audio Critic** is merely a journal of opinion—enlightened opinion, we hope—not a guardian angel protecting members of the audio community against their own hasty business judgment.*

As you know, we do issue occasional warnings about product availability (see for example Vol. 2, No. 1, p. 66, re the Cotter System 2), but our insights in this area are obviously limited, since the management practices and production schedules of a company are unlikely to be revealed by our examination of a product sample. Like everyone else, we must take the manufacturer's availability promises at face value. Somehow you appear to suggest that we ought to possess more accurate intelligence about the inner workings of manufacturing companies than do dealers, reps, distributors, export-import agents, etc. Sometimes we actually do, since we talk to a lot of insiders who tell us a lot of things, but that isn't really our line of work.

It would be generally wise to assume that a new high-end audio product under a new brand name constitutes a calculated investment risk, both for the trade and for the consumer, no matter how well the initial samples perform in the laboratory and the listening room. Audio has never been a sure-thing business like funeral parlors, but you wouldn't be in it—would you, Mike—if the fun and the potential profits didn't outweigh the risks.

—Ed.

The Audio Critic:

We read your comments about our FM-600A power amplifier in your Vol. 1, No. 6 issue and would like to give you the following information:

Our products were imported and distributed by Dayton Wright Associates Ltd. (today we do have a few selected dealers in the U.S.A.) until mid-1978. One shipment in September 1977 included the FM-600A, No. 133. All these amplifiers were tested and tuned to the same accuracy as today (see enclosed description) and left Switzerland in perfect condition. In the meantime we found that some of the amplifiers they imported did in fact ring (although we did not hear of one that was ringing 125%). This ringing is not a feedback problem as you assume, but was due to the fact that on a few amplifiers they changed the compensation of one of the amplifier stages. This could be a reason for the indicated hardness in the upper registers.

Please note that our amplifiers *do* have a certain overshoot when loading with different impedances and this is deliberately done so (e.g. as with some old Marantz amplifiers). I cannot move into theories here as this would need much more time and space. However on our amplifiers there should be only very little ringing. Therefore we expect that you received one of the slightly "modified" amplifiers. This might be interesting for some of your readers, although until today we have only received two comments on this.

. . . I thank you for being able to bring the above to your attention and remain,

Yours sincerely,
Manuel Huber
FM Acoustics Ltd.
Zollikon, Switzerland

A recheck of our laboratory measurement data on the FM-600A reveals a simple *error in computation—the overshoot was in reality exactly one half of the 125% reported. Sorry about that, but the sound of the unit was still what we heard. Incidentally, all overshoot figures reported in that particular issue of* ***The Audio Critic*** *(but not other issues) should be halved. This is of relatively small practical importance, since the duration and decay pattern of ringing have more to do with listening quality than the amplitude of the first overshoot. In any event, we genuinely regret having reviewed a corrupted version of your product and hope to make amends in the near future.*

—Ed.

* * *

The following letter constitutes something of an editorial embarrassment. We really shouldn't publish it, since malice without the saving grace of wit makes poor reading, especially when coming from someone who doesn't even have his basic facts straight. We have no choice, however; it happens to be our policy to allow reviewees to respond to our reviews in this space. So, palatable or not, here goes.

The Audio Critic:

Thank you for the kind review of the FET-5 Mark V in the most recent issues of The Audio Critic.

Unfortunately, your testing procedure is quite flawed.

A competent engineering evaluation of the Mark V would show the following results:

1. The phono section slew clips on 10,000 Hz square waves of any amplitude from amplitude clipping of the phono section down to 5 mV peak input.

2. The ratio of the low-frequency small-signal cutoff vs. power supply stiffness after regulation produces rising harmonic distortion with lowering input frequency on any amplitude of 20 Hz square wave input signal. *(Sic.)*

3. The buffer stage overshoots more than 50% on 10,000 Hz square waves in excess of 0.5 volt peak input level.

4. The buffer stage also exhibits problem #2.

5. The output IC stage slew clips on any amplitude of 10,000 Hz square waves.

Inasmuch as the problems described above produce harmonic distortion and IM distortion in the audio range, and inasmuch as we consider any distortion measureable at all to be undesirable (I hope you don't consider distortion to be desirable), the measured results predict the following sonic problems:

(1) Excess harmonic distortion in the audio range, worst at low frequencies but audible at least into the midrange. This would relate somewhat with your vague subjective description of "hooded midrange", and since I expect you have never heard audio electronics with undistorted low-frequency output, it doesn't surprise me that you failed to describe subjectively the bass problems, because until you hear undistorted electronics, I suspect you are not aware of the distortions you are always hearing.

(2) Loss of resolution at higher frequencies due to slew clipping with excess IM products throughout the audio range, which would relate to your vague subjective description of "obvious colorations."

More unfortunately, your publishing schedule is also quite flawed. The Mark V was discontinued and off the market before the review was published. Perhaps The Audio Critic should be considered to be more of a history book rather than a buying guide.

You did make one accurate statement in the review. There is a Mark VI, the FET-6 modification of the PAT-5, which since we have learned how to accurately measure and analyze the distortions we can hear, we have been able to eliminate all of the problems mentioned above. The FET-6 has zero slew-induced distortion under any condition of input, operates always within linear transconductance, has low-frequency time constants that are valid and will put out perfect square waves under any condition of input up to clipping. No phono cartridge can overload it. Of course in accordance with Jung, the slew rate and power bandwidth cannot be specified.

Since we know of no other preamplifier that will pass our engineering evaluations (except for four of our other new designs), we would predict that the FET-6 will probably be subjectively superior to all other preamp designs in existence. We would of course also predict that any preamp that does pass our test series would be at least as musical as the FET-6.

We would be happy to prove this to you and would normally offer to update the Mark V to our new standards for you at no charge. However, we understand that you sold the preamp in question which we had previously modified at no charge. That makes it more profitable to review expensive equipment if that is your policy, doesn't it?

Thus, we suggest you use the profits from the sale to acquire another PAT-5. We will modify it for our standard price of $200, including return shipping. If you are not interested, that is your problem, not ours.

By the way, your phono equalization test procedure is also flawed. Since an RIAA equalized preamp in essence drops forever at 6 dB per octave, and since you cannot build a reverse RIAA equalization circuit that rises forever, the input to test for RIAA equalization is flawed. In addition, the reverse RIAA network must act as a low-pass filter (*sic*), which will mask most phono preamp problems by eliminating high-frequency components the preamp will see in the real world.

The implications of this letter are quite obvious. Referring to the cartoon on page 6 of this issue of The Audio Critic, it appears to us that your efforts are mostly in the same league as the bearded gentleman. You cannot continue to claim superior technology in review procedures when your procedures are inadequate. You cannot expect to measure equipment under narrow-band conditions and predict results under wide-band real-world use conditions unless you live in an RF shielded room. You cannot make nonscientific statements such as in the Hafler DH-200 review, "we measured 30% overshoot on square waves . . ." when you fail to specify the frequency or amplitude of the square wave. A statement like that would get you laughed out of any really scientific journal. In the seminar published in this issue, only Matti Otala's comments were consistently within the realms of objective rationality. I suggest you take a few lessons in scientific methods from him. Understand that we don't claim to be perfect, we are not. Our argument with you is that it appears that you do claim to be perfect, and you are not. We view such claims with suspicion.

Sincerely,
Frank Van Alstine
Jensens Stereo Shop
Burnsville, MN

P.S. We will update any PAT-5 we have previously modified for no more than $100.00.

We refuse to dignify the above outpouring of undisciplined techno-rant and sheer personal bitchiness by responding to it item by item. Even if one or two of the engineering points made by Frank Van Alstine bear some distant relationship to the facts of electronics (after all, it's very difficult to be 100% wrong all the time), his newly discovered priorities in preamp design, his perception of our test procedures and his understanding of certain basic principles of physics are, to use his phrase, quite flawed. In fact, his letter is a perfect illustration of what we're talking about when we warn our readers against the semi-educated gurus operating in the twilight zones of audio.

Just one quick example—and no more!—that even the less technically minded might be able to appreciate: When reproducing the leading edge of a square wave, an amplifier doesn't know how long it will take for the trailing edge to arrive, i.e., how long the ramp will be. The amplifier has no prescience; it overshoots and rings or it doesn't, depending on the rise time of the leading edge and the various characteristics of the circuit. Thus, you don't have to specify the frequency of a square wave that makes an amplifier overshoot and ring, not-

withstanding sophomoric debate phrases like this-will-get-you-laughed-out-of whatever.

More disturbing than this howler and several others like it in Frank Van Alstine's letter is the utter predictability with which he will replace the world's greatest circuit with something even greater every few months. We defy any publication to keep up with it. The Mark IV, immediate predecessor of the Mark V we reviewed, came to us with a note saying, "You're not going to believe it!!!" And now even the Mark V is the butt of high-minded self-ridicule, like Cyrano's nose, because the Mark VI makes it totally obsolete. Were all these laws of nature still undiscovered when the Mark IV and Mark V were designed? Will the inevitable Mark VII make all of them look like a joke? We deeply regret that we took any of this nonsense seriously to begin with.

The only other thing we wish to comment on is the swinish insinuation that it was somehow greedy or financially unethical of us to sell our PAT-5, which was our unquestioned, fully paid-for property, simply because it had been diddled with and blessed by Frank Van Alstine. The truth is that we took a huge loss on it because nobody wanted it, but that's irrelevant to the case. What's relevant is that a man capable of a low-down slur like that, no matter how angry or frustrated he may be, must be deemed obviously unfit for consumption by civilized people with a sense of right and wrong. Good-bye, Mr. Van Alstine.

—Ed.

* * *

The Audio Critic:

. . . The recording of *Cantate Domino* was not—repeat, was not—made with B & K instrument mikes but with a mike called Pearl, made in Denmark.

This isn't just to nitpick at you but to let you know—because, in fact, this Pearl mike is the second best mike I have ever heard . . . The B & K's are by far the finest mikes I have ever heard. They are gorgeous-sounding things . . .

Proprius started out with the Pearl mike and switched over to B & K's—and you are correct, most of their recordings are made with B & K's but not *Cantate Domino*. I telexed Proprius, and they said two Pearl TC-4's, I believe . . .

I have been working with the two-mike technique—and only that!—for some time, and this setup is easily the finest and cleanest. I think you have heard some of these tapes made with B & K's, so you know.

. . . Your magazine is doing a very fine service by having Max Wilcox write his superb articles bringing back really the only true recording technique which captures sound as it really exists.

Sincerely,
Jonathan L. Horwich
Clearwater, FL

We stand corrected on Cantate Domino. *Actually, the Pearl condenser mike is an old familiar face to us; we owned a pair back in the mid-1960's, when hardly anybody knew about them. (An earlier model, of course.) They sounded fantastic. But they were made—and we believe still are—not in Denmark, but in the part of Sweden nearest Denmark, in the vicinity of Helsingborg. That's just across the strait from Hamlet's castle, so you weren't far off.*

—Ed.

* * *

Finally, some psychoacoustic observations by astute listeners.

The Audio Critic:

. . . I am convinced after reading many opinions that most of the differences between high-quality components are often in the listener's head or explained by certain electronic phenomena.

For instance, I feel that depth and stereo stage width as applied to how a cartridge reproduces often is due to phase problems in the cartridge. In other words anomalies are being extolled as desirable effects in high-quality gear.

Also a great deal is made of stereo imaging and placement of instruments. A concert in a large hall just does not give this unless you are sitting very close. Without visual clues one cannot pick out one instrument and place it perfectly, so this is something that really isn't true to life yet is used as a criterion for superior speakers and other equipment . . .

Sincerely,
Will J. Price, M.D.
Hyannis, MA

What your cartridge, amplification chain and speakers must reproduce accurately is not your own memories of the concert hall but, perforce, what the record producer captured with his microphones. Ideally that should be a concert hall sound you can relate to, but seldom is. Usually what the producer records is the sound you would hear in the concert hall if your ears were extended on fifty-foot antennae and aimed at the musicians from a few feet above their heads.

We, too, have often sat in the concert hall with closed eyes and observed that there was little or no imaging in the hi-fi sense. Therefore pinpoint localization is unquestionably an electronic artifact, but in our opinion more often due to the recording process than to the peculiarities of the reproducing equipment.

It also happens to be true, however, that speakers with extremely small angles of coherent radiation can give you that snap-into-focus effect as you move your listening position or even just your head. That, too, is an artifact, though much prized by some audio cultists. The ideal is a naturally recorded sonic perspective combined with proper radiation launch by the speaker.

—Ed.

The Audio Critic:

I have your first Reference B system except for the disc playback components. I have to admit that it's *far* superior to any other "popular brand" component stereo.

However, I have a problem which is not unique, from what I hear from other audiophiles. My concentration is often drawn from the music to the sound of the music. The better the system, the more I listen to the system. I have found my most enjoyment out of music itself on my car radio. Still, I would rather listen to my stereo over my car radio.

What am I doing wrong?

Russ McBride
Woodridge, IL

P.S. Please don't bring up your Reference A system unless it can drive me to work.

A man we know had a similar predicament. Although of moderate means, he married one of Hollywood's great sex symbols, a glamorous love goddess who happened to be really turned on by him and made him the envy of all men. Unfortunately, whenever he made love to her, he was unable to concentrate on his pleasure because all he could think of was his incredible good fortune and his number one rank on the erotic status scale. He ended up having an affair with one of the checkout girls at the local supermarket, who was not particularly good-looking but conventional hot stuff.

Our suggestion is that you get one of those kiddie steering wheels and hold it while you listen to your Reference B system. Maybe if you can fantasize that you are listening to your car radio . . .

—Ed.

In Your Ear


"What do you mean
it doesn't sound good?
I just put pure silver
wire in the headshell!"
TOM ACZEL

Assorted Editorial Rumblings, Rumblings and Grumblings

by Peter Aczel
Editor and Publisher

Lacking a monolithic editorial theme of any urgency this time, we use the opportunity to get in a few licks on this and that, follow up some unfinished business, and bring up a couple of things we haven't commented on before.

Out of habit and for easy reference, we are still numbering our editorial topics serially, without any implication of uninterrupted continuity from issue to issue.

* * *

31 As of this writing, our challenge in the last issue to bad-mouthers of **The Audio Critic** has gone totally unanswered. No one has come forth to engage us in a tape-recorded debate, confront us with whatever we're supposed to be doing wrong, and have the uncut and unedited transcript of the tape printed in our pages (at our expense!) for the delectation of the audio community.

Oh yes, we still get poison-pen letters; we still have reports of sneak attacks on our technical or journalistic credibility by various individuals within the trade; we still see cowardly little slurs, hit-and-run insults and heavy-handed innuendos in the journals of the cultist fringe. But—surprise, surprise!—none of these snipers will stand up and have a candid two-way discussion with us in public.

Well, the offer still stands; the rules of debate we proposed can be found in this same space in the last issue (Vol. 2, No. 1, p. 8, topic #30). Our stated policy of not responding to our detractors in any other way but this also stands (expect for published letters to the Editor, which are still subject to possible rebuttal). That means no editorial recognition and no free publicity for creeps who can't look us in the face when they talk against us.

* * *

32 To offset the bad taste left by the shrill hostility of the few dozen techno-loonies and storefront gurus who feel threatened by the "second opinion" we represent, we must also report love and kisses on a much more massive scale, involving many thousands of reasonably sane audio enthusiasts and music lovers, as well as some of the keenest professional minds of the audio world. Just recently, between the last issue and this one, judging from our subscription renewals, new subscriptions, love letters from subscribers (which we don't publish unless they contain editorially interesting information) and press notices, our acceptance has taken a quantum jump. We're finally beginning to feel that the difference between our equipment evaluations and the available alternatives is becoming fairly common knowledge, not just the perception of a few insiders who always knew where we were coming from.

Of course, the support of these technologically sophisticated insiders remains our most precious asset. In a recent newspaper interview, we stated that one of the important differences between **The Audio Critic** and other noncommercial or underground audiophile reviews is that we appear to have *some* credibility even among top technologists and academicians who know a lot more than the Editor, not just among rank-and-file audiophiles who know less. We added that we doubt very much whether the other editors can make such a statement—if they are honest with themselves.

We find it particularly gratifying, however, that those rank-and-file consumers of audio equipment also seem to sense and appreciate that at this point we have more to lose than other publications if we turn out to be dead wrong about something and that we therefore try to make damn sure we're right, especially about the big ones. When you get right down to it, it's extremely difficult for such a consumer—a music-loving history teacher, let's say—to choose between a favorable and an unfavorable review of the same piece of equipment by two different reviewers. He can't possibly judge the *total* technical and psychological discipline, or lack thereof, that went into each conclusion. Somehow he goes by the overall vibrations—self-esteem or self-importance, scientific aura, seriousness, vocabulary, etc.—communicated by the reviewer, plus past experience with the latter's recommendations. All very unsatisfactory, since some quacks are pretty impressive and some reliable practitioners less so, and since no one likes to admit that he fell for poor advice on a previous occasion and bought a piece of junk. What we're trying to say is—we have, and always will have, a communications problem in this area, but as it turns out we could be doing a lot worse than we are.

* * *

33 Some of the favorable comments received from our subscribers are mixed with reservations to the effect that (a) we are much too technical and fail to explain audio engineering terms we use, or (b) we aren't technical enough and fail to be scientifically rigorous.

We'd like to respond to that by reminding everyone that **The Audio Critic** was conceived for and is primarily addressed to the *consumer*—a very special kind of consumer, to be sure, who wants to know the rationales and proofs, not just the product ratings. What **The Audio Critic** emphatically is *not* is either "My First Book of Electricity" or, on the other hand, a doctoral thesis. If you want to know the difference between resistance, impedance and reactance, any good encyclopedia, technical dictionary and even some pop-tech newsstand magazines will give you the answer. (*The Wall Street Journal* doesn't explain, either, the difference between a stock split, a stock dividend and a spin-off.) Nor can we be expected to test and review literally hundreds of pieces of equipment with the same kind of academic rigor as a scientist brings to the few dozen research papers of a lifetime—or, for that matter, as a research-oriented manufacturer can apply to one or two products a year.

What we have to offer is nothing more than (1) a very nicely equipped laboratory, (2) a correctly aligned reference stereo system with exceptionally high resolving power, constantly upgraded, (3) an attitude informed by the laws of physics and the love of music, and (4) a little more time than the typical audiophile has available for judging any single piece of equipment—but just a little more. That's all. We don't teach kindergarten and we don't tell PhD's how to do their job. And we certainly don't claim to be the sole possessors of any piece of knowledge about electronics or electroacoustics. But when we say that speaker A is more accurate and sounds more like music than speaker B, we make sure that neither scientists nor hairdressers can possibly misunderstand what we mean and why. Their money is equally good, and our job is not to allow them to waste it.

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34 There also seem to exist some strange ideas in certain audio circles as to just what constitutes "scientific" audio journalism and reviewing. Diagrams, graphs and numbers, no matter how fatuously conceived and irrelevant, are the irrefutable credentials of the scientific tester in this crowd, with special emphasis on very small and very large numbers. Especially favored are extremely complicated explanations of the simplest things, preferably with newly coined buzzwords.

Since we believe that science is essentially a highly developed and mathematically implemented form of common sense, we want to alert you to the dangers of this insidious form of charlatanism, which perverts the sophisticated consumer's aforementioned desire for rationales and proofs. A snake oil promoter is still selling you snake oil even if he shows you a fat book with cross-sectional drawings of the snake's oil-producing tissues, tables of snake oil viscosity indexes, and diagrams of the snake's brain waves. Whereas a shirt-sleeve technologist without phony pretensions may casually remark that a line-contact stylus reads a larger and more representative sample of the information on the groove wall than a spherical stylus, and right there he has told you something quite profound, practical and scientific.

* * *

35 Neither science nor honesty nor even experience is sufficient, however, to keep the reviewer out of trouble at all times in the delirious world of consumer audio. Take our recent recommendations of the Rappaport AMP-1 and Hafler DH-200 power amplifiers, for example. Both have turned out to be a credibility problem for us.

In the case of Andy Rappaport, how could we possibly have predicted that with all his initial success and rapidly rising sales he wouldn't turn out to be enough of a businessman to stay even marginally solvent and keep out of bankruptcy? Now there are all those marvelous-sounding but painfully touchy AMP-1's out there, with no more Rappaport company to service them under the original three-year warranty. (See the power amp article in this issue for possible help.) Or how could we have predicted that the designer of the DH-200 would leave the Hafler company and that they would then produce slightly variant versions, none of which sounded as good as the one we had originally reviewed? (Until very recently, that is; again, see the power amplifier article.)

A totally effective and reliable consumer-oriented testing service would have to maintain a network of inspectors to check out the business management as well as the production lines of each audio company whose products are under review. Fat chance. We don't even have the staff to step up our publishing schedule, let alone to police the industry. Nor does any other publication. The commercial hi-fi magazines get around this problem fairly consistently by reviewing only the products of well-established companies that advertise in their pages. That way, of course, they miss some of the most original and most exciting new products, which are almost invariably made by the Rappaport type of company.

If you have any idea how we could serve you better in this very difficult area, please let us know. We don't see how we can spot management-related future problems in an excellent and well-behaved product submitted to us, but maybe you can tell us something we haven't thought of.

The Audio Critic Seminar on the State of the Art: Part II

This is the second and concluding installment of the edited transcript of our 8-man, 15-hour techno-talkathon, already a minor legend in its own time. Of course, to get the most out of it, you ought to read Part I first.

The background and circumstances of our seminar, held early in 1979, were explained in detail in the preamble to Part I in the last issue (Vol. 2, No. 1). Minibiographies of the eight participants were also given, so that none of this information needs to be repeated here, especially since we want all readers of Part II to have some previous familiarity with the contents of Part I.

Comment is due, however, on the absence of the eagerly awaited discussion of loudspeaker theory and design, which we decided after prolonged soul-searching to omit from the published transcript. Quite frankly, this final section of the seminar failed to come up to the level set by the preceding dialogues. A number of interesting ideas were presented, which we plan to draw upon editorially in future issues, but by and large the discussion was unexpectedly diffuse and inconclusive, with everyone bogged down in his own definitions, goals and values, and with none of the synergistic convergence toward a shared element of understanding that had characterized the seminar up to that point. Probably we were all tired, as it was late in the evening and midnight by the time we stopped.

At the current publishing costs per page, it doesn't seem to make much sense to print this material, especially on top of the large amount of solid information about loudspeakers in every issue of **The Audio Critic**. Even so, we feel that the signal path from stylus tip to speaker terminals was explored in a uniquely illuminating way by the seminar, for which the less productive time spent on speakers was a very small price to pay.

As in Part I, asterisks (* * *) in the transcript indicate omitted sections, which are very brief and unimportant except for the loudspeaker discussion. The opening of Part II picks up the continuity exactly where we left it off, without any omission.

EDITOR: Let's try to put together a recipe for state-of-the-art phono reproduction. Let's see what we can agree on. We begin . . . Do we all agree on a line-contact type stylus? Is there anyone here who would oppose in the light of these previous considerations a line-contact type of stylus?

COTTER: I would add I like it and I want to use it, but it marries us to a pair of constraints. If you go to a line-contact sylus, you have to realize that the cutting stylus is almost always precisely perpendicular; that if it deviates, and those of us who have experience with cutting know that if you deviate a little bit one way or the other, the stylus either digs in or it plows up and you get a noisy groove. So you're sort of naturally constrained, like when you cut with a lathe, to have that sylus pretty damn perpendicular.

HEGEMAN: You don't want a gray thread coming off.

COTTER: You don't want a gray thread. As a matter of fact, that means that the cutting face, the cutting facet and the cutting line of a cutting sylus—even though there is this vertical angle effect which moves the stylus in effect back and forth, that is down the groove as it moves—your aperture is lined up. If you have a line contact stylus, you're going to have an aperture azimuthal alignment problem not unlike the gap in a tape head, in which the need to maintain that exact alignment is going to be a function of the ratio of the length down the track, that is the gap length, to the gap width in the track width sense. If that line contact is of the order of an aspect ratio of say 10 to 1, then one sees a rather shockingly small angle as tolerable before there's a significant change in the aperture. For the same reason you have to do azimuth alignment on a tape.

EDITOR: A gentleman from the University of California, I believe, pointed this out in a letter to us just recently. He pointed out that even though we're very strong for correct VTA alignment, as he pointed out correct VTA alignment may be incompatible with maintaining the perpendicularity of the stylus itself.

COTTER: If the pickup you happen to have doesn't have its effective vertical angle in exactly the correct position for that particular record, when the stylus is aligned in the vertical direction, then to get the correct reproduction you're going to have to do something else other than move the pickup up and down, unless you will tolerate some significant loss in the effective high-frequency aperture. And that's a truth, a geometric truth; it has nothing to do with size. So you're on the horns of a real dilemma, and one is forced to take another approach. I think there are some approaches, and we're pursuing that. But the fact is that the need for a line-contact stylus has been largely ignored, and it was invented for all the wrong reasons. It was invented for CD-4. No one ever looked at the principle of playing in effect the wide-track recording with a narrow-track head.

EDITOR: From the practical point of view then, since we have to live with this tradeoff between correct VTA and correct aperture, how should we proceed? Should we still get a line-contact stylus?

COTTER: Yes.

EDITOR: And should we still try to align for VTA by ear?

COTTER: That depends on which records, I think, because there are records that are

available whose vertical angles are extremely different. In that case . . .
EDITOR: All right, on modern records that are say in the 15 to 18 degree range, you would still do what we here have been doing for about a year now?
COTTER: I would say that's the best strategy at the moment.
RAPPAPORT: But it's certainly not the optimal strategy, though, because you can really deceive yourself. Because if you have a record that's recorded poorly, for instance, and it sounds somewhat muffled, it's very very tempting to lift the back of the tone arm just a little bit and maybe aggravate the front end of your phono preamp if you're speeding up the signals getting in there and just add a little bit of . . . You have to be very careful, because if you're not careful you can really deceive yourself.
COTTER: What we found was, though, that the correct angle produces a fairly less sensitive null in these effects than you would have otherwise considered had you not had a preamp that was not vulnerable to these very small mistracking effects in the sense that Matti has defined it, as the first, second, third, nth rates following . . .
ZAYDE: It sounds like an avalanche breakdown, in lieu of a better expression. You know when you're there, but if you exceed it it isn't something that goes out the window.
RAPPAPORT: Right, well as you said, that's a function of the quality of the preamp.
COTTER: Not even quality. I think, you see, we're facing an interesting problem. We're now defining a set of requirements, a set of constraints for that preamplifier that had not been considered part of the design problem before.
RAPPAPORT: When I say quality, that's quality in the sense as it relates to the job that it has to perform.
COTTER: But fairly to the other designers that exist in the world, they pursued an objective that didn't contain these definitions. So their designs are in most cases, let's say today, probably optimal for the set of constraints. The problem is that those approaches are incorrect. We're talking now about a really substantial change in direction, in which certain topological and certain system requirements now appear as a requirement that didn't appear before. So we're saying—I don't like the use of the word quality, I think it's a question of attitude or, you used the word before, philosophy. We're talking now about a change in the philosophy of the requirement, and therefore designs could change and should change.
EDITOR: Let's follow through now. We have a line-contact stylus; obviously we want a stylus cantilever that's fairly lossy but stiff at the same time, which is a problem. But this can be accomplished, I suppose. Now what about the generator mechanism?
COTTER: What about the VTA?
EDITOR: We have discussed the VTA.
COTTER: What about the vertical tracking force?
EDITOR: As large as possible, in the light of what we've said?
COTTER: I don't think one should run home and put a flatiron on the pickup, but I think . . .

HEGEMAN: How about playing it with an old Western Electric 9D, for instance?
COTTER: It's interesting. As a matter of fact, talk about that for a minute. Those were rather astonishing, weren't they? They sounded awfully good; the grooves weren't that much bigger, and talk about the forces! That's an interesting . . . You used those.
HEGEMAN: We used to call it the 5-pound monster. It was on an 18-inch arm, so forth and so on . . .
COTTER: Vertical pickup.
HEGEMAN: It was used for either vertical or lateral on that early stuff. The interesting thing was that the stylus was vertical. One of the requirements they had to have for use in the studios and so forth was this backwards and forwards cueing for the early disc jockeys, where they had to be able to cue up to something. For that reason you had to rock the record back and forth. Those things were pretty heavy. You had a warped record on it, you had a good chance of getting the stylus locking into the groove, or just taking the groove out of there completely. But it was a very—to me, at least—very fantastic sound.
EDITOR: Was that a moving coil?
HEGEMAN: Oh yes. Vertical-lateral. It had the coils at the angles so you could series-parallel them, buck them either way, so you'd get a vertical or a lateral presentation and the other component cancelled out. But the thing that was interesting to me was that they did all their equalization right out of the cartridge. They had all kinds of—there were probably 8 or 10 positions on that equalizing switch to allow for playing 78's, 33's, lateral, vertical, all this kind of garbage.

". . . nature made it simpler to get us the orthogonality we want with the moving coil, and to get more power from a given mechanical impedance."

EDITOR: They were able to throw away all that signal and still have some left?
HEGEMAN: It was a big cartridge. I wish we could work with those output levels now, I'll tell you. That was their full equalization; all they needed was basically flat gain after that. So that a lot of the electronic problems . . .
COTTER: Microphone amplifier was used.
HEGEMAN: Yeah, they used a microphone mixer, one position . . .
RAPPAPORT: Very interesting, because I hadn't heard of that, and I have just designed a preamp, in fact I've applied for a patent on a technique to equalize right at the cartridge output.
HEGEMAN: Well, I'm sorry fella, somebody got there first, a long time ago.
RAPPAPORT: So much for my good idea.
HEGEMAN: But the interesting thing, of course,is that by doing the equalization at that point, you came out with essentially a flat signal to amplify, and you did not build in some of these crazy effects that we've been talking about.

COTTER: You escaped completely all of these problems, and you still had the scanning problems, but by virtue of the enormous mass and the kinds of geometry relationships, in effect, those pickups had much fewer of the problems we're describing, of ultrasonic behavior. What's interesting is, those damn things lasted pretty long, playing records.
FUTTERMAN: I remember a cartridge, I think it was made in England . . .
COTTER: At 30 grams, right? The 6A was 30 grams, I think there was . . .
HEGEMAN: The 9D, I think, was running somewhere around 45 grams weight, measured almost in ounces instead of grams.
COTTER: 3-mil stylus, but scaled down, this is . . .
EDITOR: Is there anyone here who feels that stationary-coil cartridges have advantages over moving-coil cartridges? Moving magnet, moving iron, variable reluctance types?
OTALA: One comment, of course. With the present embodiments of these principles, I don't think there is anybody who would prefer moving magnet. But the present embodiments are not necessarily the only possible ones.
COTTER: Except that I'd say if you were looking for principles where nature gave you grace, it would certainly be in the moving coil direction, because you've got that orthogonality inherent in the generator system and you'd have to make an awfully . . .
HEGEMAN: I would say that they had one advantage, Pete. They're a damn sight higher output.
COTTER: Ha, I want to attack that.
EDITOR: We're going to come to an interesting point.
COTTER: Let me start by saying something that's interesting. Every pickup, whether it's moving field, moving magnet, moving iron, moving coil, moving zabbas, capacitance or . . .
EDITOR: How do you spell that?
COTTER: Oh, that's an Altarian technique for playing phonograph records. But they all have exactly the same noise, because the noise that they have is the thermodynamic noise. It's the basic Johnson noise. Pickups do not differ in their noise power. If they did, we could connect the better one to the poorer one at room temperature and neatly violate the second law of thermodynamics, by transferring energy from one body at the same temperature to another body at the same temperature. The fact is they all have exactly the same noise power. And it's noise power which is the thing that determines signal-to-noise ratio ultimately. The way in which they differ is in their ability to abstract energy from the movement. Now you're talking about a sort of electromagnetic generator efficacy. I submit that if you look at the existing designs and you also look at the fundamentals, it would appear very much easier to get more power from a given mechanical impedance at the stylus with a moving coil principle—moving coil, stationary field—than it would with a moving field approach. In fact, when you look at the results, present-day moving coils differ by anywhere from 10 dB in the poorest cases to as much as 35 or 38 dB greater power than the moving field versions, which are disguised because of the difference in impe-

dance levels. It isn't voltage, it's voltage times current. All the things we've talked about before, including the newer understanding of what signal-to-noise ratio comes from in a pickup, suggest that it would be a worthwhile pursuit to get a better signal-to-noise ratio inherent in the generator system of the pickup so as to be able to realize some of these improvements.

HEGEMAN: I didn't say impedance matching; I just said you get higher voltage output out of magnetics.

OTALA: Take a typical example, though, of unused possibilities. Nobody has, as far as I know, tried to use the transductor type of moving magnetic . . .

COTTER: Parametrics.

OTALA: Well, you could easily do that, and that would give you energy as much . . .

COTTER: It's just an alternative amplifier.

OTALA: That's an alternative amplifier, but probably with all the advantages of a moving coil.

COTTER: Yes. If you made a pickup amplifier that was in effect a parametric system, you would have a lower than room temperature noise system if you chose to do it in some particular way. There are certain kinds of conditions and restraints there with respect to the one-wayness of the amplifier.

EDITOR: That still wouldn't eliminate the Rabinow-Codier effect, would it?

COTTER: No, what Matti is saying is that one could build a lower noise preamplifier, or make a pickup perhaps that even contained the amplifier mechanism if you used a parametric approach. What we're both saying is that that wouldn't alter the basic transducer relationship, in which getting a lot of power still has corresponding advantages. So actually these principles apply even to the improved technique. I think that's the important idea. So what we're saying is that if you were starting with the easiest approach, it would seem as though nature made it simpler to get us the orthogonality we want with the moving coil, and to get more power from a given mechanical impedance.

EDITOR: Basically, then, are we all agreed that line-contact stylus, well-damped but stiff cantilever, and moving-coil generator are the way to go today?

OTALA: Who knows? Who knows?

HEGEMAN: By popular demand, I would agree to . . .

EDITOR: My readers want definite answers, none of this highfalutin academic theorizing.

WILCOX: Don't let the facts get in your way.

OTALA: Well, that is the present choice, and that is what you asked about the state of the art today. But we didn't answer the state of the art in the future.

RAPPAPORT: That's right. There's no point in discussing the state of the art today; it's a question of how we're going to get to the state of the art tomorrow.

HEGEMAN: I'm the state of the art tomorrow! You don't know me!

RAPPAPORT: I just hope it's different from today, that's all.

EDITOR: I suggest we track through the possibilities through the whole phono system. Let's get the signal up to line level and then let's go back. We still have to cover tone arms and turntables, which are the passive components in this, but since we're discussing the electrical signal itself let's follow it through then. We're now at the crucial juncture of choosing a transformer, a head amp, right? Aren't we?

FUTTERMAN: Pre-preamp.

COTTER: Now we have the problem: how do we get the thing amplified? In what way is the problem different than in the flat amplifier?

EDITOR: Obviously if there's enough power coming out of a moving-coil cartridge, what you need is simply an impedance transformation. The question is how do you accomplish that in the best possible way? I know that there are some differing opinions around this table on that very subject. So let's hear them.

COTTER: I have a bias. It says the simple thing to do is just transform the energy through a transformer.

OTALA: I have the same opinion, not because that would be an elegant solution necessarily or that it would be easy to make a transformer of that kind, especially for that amount of octaves. But having tried almost every preamplifier worth trying in the first place, I mean pre-preamplifier, and coming to the fact that the transformer type of pre-preamplifier had the lowest psychoacoustic masking, then I tend to prefer the transformer as being state of the art. But suppose, somebody goes and invents a pre-pre-amplifier which is capable of doing the job. In the future we probably will see that kind of amplifier. How this is done, that's another thing.

RAPPAPORT: Actually, there's no need at all for a pre-preamplifier. There ideally should exist no pre-preamplifier. If we accept the existence of the moving-coil cartridge as being the state of the art today, then the idea is to design the entire equalization and preamplification system around that cartridge, and there's no need for a transformer or an electronic flat amplifier to step up the voltage. A better approach might be to combine the equalization with the step-up and just deal with a preamplifier with a sensitivity of roughly 20 dB greater than our standard magnetic preamplifier.

HEGEMAN: Commerically, that's a very difficult one, Andy.

RAPPAPORT: I've been trying it for years and you're right; commercially it's very difficult.

COTTER: The thing is, you've got a problem which comes about in audio systems—maybe it bears a little notice at this point—that actually makes the problem for the designer of the elements extremely complex, because of the lack of definition of interfaces—impedance, power, voltage levels and so on. We deal with the world of wild and variant kinds of interfaces, and that is certainly a nonoptimum . . .

OTALA: The good old thing of having a high-sensitivity piezoelectric type of interface—one volt, and that's it. Out of the pickup.

HEGEMAN: Isn't that what that strain gauge pickup produces approximately, or something like that?

OTALA: It has got an amplifier inside to do that.

EDITOR: The whole thing is an amplifier.

RAPPAPORT: That's right. It's just a variable resistance which we put into an amplifier.

EDITOR: The thing is then, we have three alternatives. Put the moving-coil signal through a transformer that brings the voltage up to the level of a typical moving magnet or moving iron type of pickup—that's one possibility. The other is, put it through a flat amplifier that does the same; or, three, put it through an equalized preamplifier that brings it directly up to line level.

COTTER: But there's a consideration here of noise figure. Optimization, minimization of noise in the system. It would seem, for a variety of reasons, that it's easier to match a somewhat higher impedance level to get an optimum noise figure than it is at a very low impedance level. That's both a device and a circuit design problem. And there's the question of cost. In the ultimate, the noise injected by an amplifier will always be larger than the noise injected by a transformer. The path I chose at the time I chose it was to go the transformer route because it looked like a better path.

OTALA: Well, you say the ultimate is always greater. I see no basic physical reason for that.

RAPPAPORT: We don't have devices yet which will allow us to . . . well, we could use hundreds of them in parallel. Theoretically it could be done.

OTALA: Yes, this is completely true, but it doesn't mean that we wouldn't have that kind of devices tomorrow.

RAPPAPORT: That's right. There's no reason why they can't be built.

COTTER: You do have a very practical problem in the amount of current that flows in your amplifying path; as long as it has that one-way electronic emission, you're going to be governed by the shot noise equation and the perviance equation, both of which are limited by materials and geometry. Even if you take the Richardson equation, which is the field emission situation, which is *the* most advantageous, that still gives you a very very large current that you have to have in order to deal with the 3 or 4 or 5 or 6-ohm source impedance. It seems as though that is not only uneconomic but leads to some practical problems building semiconductors. That is, the people who've tried to do this wind up with either very large or a very large number of devices which becomes very uneconomic. Or when you try to get into a small device, you start . . .

RAPPAPORT: When you speak about signal-to-noise ratio, though, the idea is that our limit is the signal-to-noise ratio in the groove, primarily. In other words, what is that signal-to-noise ratio, and then . . .

COTTER: Ah, but we just talked about that, and we don't know what that is.

RAPPAPORT: But the point is if that signal-to-noise ratio is equivalent to say . . . with a moving-coil cartridge the signal-to-noise ratio is normally measured with a 1-millivolt signal . . .

COTTER: You're saying that's a 15 dB noise figure, why fight for a ½ dB noise figure.

RAPPAPORT: Exactly.

COTTER: I quite agree. The point that I think is important to make is that we don't know what the noise level of a disk really is yet. And we'd better start looking for it. But

we're going to look for it with these line-contact styli with larger vertical forces, and we're going to do something about understanding what happens when we make a record to give us a noise level, and we may discover that we want another 20 dB.
OTALA: Coming back to another thing, I don't think for instance that using enough devices in parallel would be any problem. We've got those devices—the National super-matched-pair transistor, for instance. Just IC technology—how many would you want to connect in parallel? That's your choice.
COTTER: You keep running the collector current up.
OTALA: Right, that's true. If you don't do that you're having another horrendous problem, and that is if you decrease the unit collector current, the f_T goes rather rapidly down. And it's a steep function of current at that level. So you're going to get a large variation of f_T, and consequently, since you are taking gain from that stage, you're then also causing phase modulation.
COTTER: Another view of the problem is that you're dealing basically with a current generator—we're talking about a low-impedance generator—and amplification is equivalent to saying you're going to have many times larger current output which becomes these rather humongous currents. And still at a fairly low impedance level. So this is a question of economics as well as a question of practicality. In the long run, you can build a transformer—if you assume you have theoretically no limit to the size, and to a certain extent you do have that virtue—you can make a transformer of vanishingly small resistance. To make an amplifier of vanishingly small equivalent noise resistance, like ½ ohm or something, gets to be a pretty large ensemble of conductors whose surface area becomes . . .
OTALA: Making vanishingly small resistance transformers is also quite a problem because you're losing coupling then, in that case. And furthermore, the problem there is that, even if you would be able to do that, then you're still basically limited to, let's say, four decades, five decades of frequency of proper operation due to various reactances.
COTTER: No. I think it's possible to make much more range than that. We've done it. But I find it very hard to build a preamp of comparable performance using what I know is available in the way of devices.
OTALA: You're exactly right; that's what I said too. Today the transformer seems to be the only alternative but not necessarily, I believe, tomorrow.
COTTER: Interestingly enough, when you go to the high currents, you wind up with a very significant current loop condition, which means you have to magnetically shield the system rather significantly because the current loop becomes a magnetic antenna. You wind up with shielding requirements. We just opted for that because it seemed that in the long run, the opportunities provided by nature at the time—and perhaps even into the future, if you look at what semiconductors can do—we said with the available emissions didn't seem likely. So we went that road.
OTALA: The transformer has a very good property, though, and that is the filtering effect.
COTTER: Ah, you could make it anything you want.
HEGEMAN: Unfortunately, I'll have to vote a little in the other direction, because I have never yet heard a transformer in which I don't hear a change in the phase relationship of the upper harmonics of the violin, which acts as a constriction and a hardening element, as far as the string tone is concerned. Now I admit I have not had any opportunity to work with Mitch's transformer, but this goes back to the early days, the early mono days with various cartridges.
OTALA: You know how many transformers you have in a signal path before it's cut into the record?
HEGEMAN: You know what I listen to? I listen to recordings made with no transformers—for my strings—so I have perhaps a different perspective.
OTALA: Take almost any commercial mixing console; there's at least six transformers through which the signal . . .
COTTER: Most any microphone these days has a transformer.
OTALA: So that signal is already so contaminated.
HEGEMAN: I know, so why should we contaminate it any more?
OTALA: Yeah, well, there's something else coming. You know all that talk about Dolby records, and things like that.

". . . almost invariably, when the distortion levels were small or reasonably small, the subjects picked the distorted channel as being the original."

HEGEMAN: To me that's unbearable contamination.
OTALA: That's true, but you see, the problem is that the more we get advanced in the technology, and the more we are aware of the contamination, the more we seem to start to contaminate. It was not long ago when they invented the noise reduction systems. It was not long ago when they started playing with . . .
HEGEMAN: 24-channel mixdowns.
OTALA: Okay. It's not long ago when they put all those LM-301's into the mixing consoles. It is not long ago when they started to use Kepexes and other units to spoil the rest of the recording. It was not long ago when they started to pre-distort the recording. And heaven knows—it's not more than four, no five years ago when they invented the dynamic limiters to reduce the high-frequency signal content, and they act exactly as TIM generators. Now for heaven's sake, this all has happened during this decade, and we are supposed to know something about sound quality!
COTTER: It's all proved as not important, because you can't hear the difference.
HEGEMAN: If you don't get a chance to hear the difference, you can't hear the difference. I agree with that.
RAPPAPORT: The problem is, we have two paralleling technologies. One is on one end of the stick, and they're going towards ease of production, and to a certain extent gimmickry, but it's allegedly with a purpose. And then on the other side of the coin, you have us here, and we're trying to purify . . .
COTTER: Fidelitists.
RAPPAPORT: Yes, the fidelitists. Exactly. And they're two technologies which parallel but never intersect. And it's very unfortunate.
HEGEMAN: I don't consider them parallel, Andy; I think this is a divergent situation.
OTALA: But since this is a commercial world, and since we know that the percentage of unpolluted records, for instance, is decreasing rather than increasing, aren't we fighting a losing battle? I think we are.
WILCOX: I don't think that's really true, no.
HEGEMAN: I don't agree with Matti on that. Let's keep on with the battle.
COTTER: Let me sound another note of optimism. Max is the one with the real note of optimism.
WILCOX: I think that it's changed. I don't know that this is the time you want to talk about it.
EDITOR: The very fact that this publication exists and is growing seems to indicate that there is some interest in the purist approach.
RAPPAPORT: The interesting thing is that every time we are allowed to listen to a superior component, we realize exactly how much better records are than we thought. And they've been standing up, as Mitch says all the time, they've been standing up very well to the recent onslaught of improved components.
COTTER: I don't think we've ever played a record yet. I've gone around saying that, and I think there's probably more still locked up in the record than we are capable of extracting, that's good music.
EDITOR: Maybe this is not the moment to interject this, but this will soon stop when they switch to some kind of cockamamie digital process with a sampling rate of fifty thousand.
COTTER: I have an abiding faith it will not happen the way they claim.
RAPPAPORT & HEGEMAN: I hope not.
EDITOR: I would like to take this up later on; let's not forget about it.
COTTER: The fact is that there already exist more than a quarter of a million LP titles of incomparable majesty. When you look at the range and the character of music that is available to anyone today, no one in the past ever had access to that kind of wealth, be he king or prince or gold merchant of the universe. You can't possibly imagine the sort of resource that that made available. Somebody at the age of 20 can have acquired more musical exposure than Mozart, Beethoven, Brahms and Telemann rolled into one. The number of hours of variegated music listening that this makes possible, however imperfect it is, is still an adequate stimulus to give you an exposure that you never had available.
OTALA: Which only goes to prove one thing. When we did our psychoacoustic experiments, we came to the staggering result that almost invariably, when the distortion levels were small or reasonably small, the subjects picked the distorted channel as being the original. That only means that we have become acquainted with

distortion sources. Very few people go to concerts. They listen to radio and records.
COTTER: I was talking at lunch about the fact that the first 55 years of the phonograph history were spent listening to mechanically-acoustically recorded records, reproduced acoustically-mechanically. And that we grew several generations of people whose identity with the record and the recording art was synonymous with distortion. Nobody doubted that it was distortion.
EDITOR: This is characteristic of many aspects of today's plastic culture.
COTTER: Orange juice.
EDITOR: Orange juice, coffee. Can you imagine—I always bring this up as an analogy—that hundreds of years ago, coffee beans were brought into Western Europe, and coffee drinking swept the Western world. Of course the original Arab way of doing it was to take the green coffee beans, roast them on an open fire then and there, grind them then and there, and then create a brew. And it was this magical brew that swept Europe, I believe in the 16th or 17th century. Can you imagine if it had been a can of instant coffee that was introduced at that point, that it would have swept Europe?
COTTER: On the floor, not in the cup.
EDITOR: The analogy goes for music.
FUTTERMAN: But I remember when, in the early days of radio, everything broadcast was live, at least 90% of it. So we listened to live music, even though the components . . .
HEGEMAN: It was not live music, it was live performance.
EDITOR: Even in the early days of FM, Major Armstrong's station in Alpine, New Jersey, had lots of live broadcasts.
OTALA: But this is not what we are pointing out. It was live performance, which has nothing to do with live music itself. Live music is an acoustic sensation and if it is passed through a channel which has distortion, and if that's the only medium that people listen to, they get accustomed to the distortion, and they start thinking that that is the original. After we had these funny problems with people in the psychoacoustic experiments, we recommended them to go to a number of concerts. And they went. We later asked what did they like at the concert. They said the tone quality was not very good, indeed, some kind of luster or brilliance missing—how can they ever sound that bad?
EDITOR: Of course this is a personal aesthetic that you can't really argue with. The plastic experience may have a sensory value that's incomparably better than the real experience.
HEGEMAN: I never come out of a live concert, as I walk out through the doors of the hall and down the steps—I say, that just set the cause of high fidelity back another ten years.
COTTER: Would you agree, Stew, though—that the general impression of recordings is that they are over-bright compared to live performances?
OTALA: There's some kind of glitter . . .
HEGEMAN: Yes. They're overbright; they don't have the spatial, airy characteristic that a live performance has.
WILCOX: Yes, but there are various obvious reasons for that.
COTTER: Talk to me.
WILCOX: The obvious reason was the invention of the condenser microphone by—I don't know if it was Gerhard Neumann who invented it, but anyway . . .
COTTER: It's a very, very old format.
WILCOX: I remember working with engineers who went from the days of the old RCA 44, the ribbon microphone, which was a flawed device but within its frequency range a pretty smooth-sounding device. Then the condenser came along, and it came along with its pre-emphasis in the high end, and with the cardioid pickup, which also gave you more direct signal and less reflection. So you got two things. You got a brighter sounding microphone, and you also got a more direct sound. All of that added up to an edgy kind of sound.
COTTER: Why was it accepted?
WILCOX: Because it was supposed to be brilliant.
COTTER: Speakers I think, Max, were rather poor . . .
OTALA: There's another thing to this. When Armstrong experimented with FM radio here, he was particularly proud of the high-frequency response, and that was boosted—and that created that sh-ch-ch type of sound.
HEGEMAN: I have recordings of a couple of his Army band concerts that came up from Washington, DC over his special 15-kHz line using Western Electric 640AA mikes. They are very brilliant. It was very close-miked, and it shouldn't have been. The 640 is not a mike you can close-mike with.
OTALA: This established some kind of standard synonymous of good sound.
WILCOX: I was an 18-year-old college kid, and I decided I was going to work for the summer and buy hi-fi equipment. So I went to Chicago to a place called Allied Radio, to their sound place. And the sound that was coming out of that place was so horrendous.
OTALA: Well, it is horrendous if you take a normal receiver and try to tune to New York stations.

* * *

EDITOR: Could we get back to preamps?
OTALA: We have stretched his patience.
COTTER: There are no easy answers. Do we all agree on that?
WILCOX: Peter, I think you just touched rather peripherally, and then zipped out of, something that's terribly important, that I would like to talk about for two minutes. People say, why are things so brilliant-sounding in the name of high fidelity? I was telling a couple of them here. I remember deciding I wanted to buy a hi-fi system. I was 19, 18 years old, and I was a pianist; I was in college in music; I knew what music sounded like as most of my experience in music was live. I was playing violin sonatas with people, stuff like that. So I went to Allied Radio. There was a Stevens, Stephens, I don't know how . . .
COTTER: Stevens.
WILCOX: Stevens Silver Coil, or something like that, loudspeaker, and there was the new GE variable reluctance pickup that didn't have any kind of preamp yet; it was just the naked output of the thing. I went into what was their audio salon. The relation of that to live music was nothing. It was the most screechy, horrendous . . .
EDITOR: It was all highs. If the variable reluctance cartridge wasn't equalized, all you got was highs.
WILCOX: That's right. And then the loudspeakers were all tipped up. So from the Capeharts and Magnavoxes of the '30's and the early '40's, suddenly we came into the era of "high fidelity." And those of us who are old enough to remember that . . .
HEGEMAN: Those big old Capeharts weren't so bad, I'll tell you.
WILCOX: That was a very ugly period in the history of reproduced musir. And I think only now are we starting to get away from that. Most recordings are still made with tipped-up microphones. They sound like very wide-range "transparent" recordings. I would have to differ with you; there are some wonderful recordings; there are also some recordings which sound incredibly ugly when you play them on really wide-range equipment because you see how screechy and awful-sounding they really are.
COTTER: But some of those older recordings in spite of their tipped-up responses have a cleanness due to their basic simplicity.
WILCOX: That's right. Because there are not so many microphones involved.
OTALA: Some of those recordings, however, also show another thing. With some of the cutters that are not really exactly the best, and some of the second-generation mixing consoles, we found a couple of records which by masking evaluation had about 35% TIM. 35% rms continuously is quite a number. You can't imagine!
COTTER: Coming back to this high fidelity phrase, I would like to introduce another idea. It seems to me that high fidelity came to mean something really very opposite from what high fidelity is supposed to be all about. Since we came together today to discuss state of the art in sound or whatever, but it seems to me that what we're talking about is largely something in the service of music. The idea was musical sound, musical ideas. It struck me that the phrase "high fidelity" closely resembles the phrase "painless dentistry." The minute the word "painless" appears, you know you have something to worry about. "High fidelity" implies a certain struggle, and I think what we ought to make clear—because I think there are a lot of people who say "Oh I know about high fidelity, I've heard high fidelity, it doesn't sound like music, it's too loud, and I really can't stand those high frequencies." And that's what it means—it means almost too much pain for most people. I think what we're all talking about is something very different from that concept of high fidelity. What we're talking about in fact is a kind of attainment, in which height no longer has any meaning because it's high enough. So we're really talking about music. And if we're talking about music, we're talking about unmusicality and musicality. It is possible, I think, to get a system that is less than perfectly accurate but which is more musical than other systems that blindly and foolishly pursue accuracy without assessing the level of pain that is produced. I wonder if we can talk a moment about that aspect, because it comes to focus in records and in preamps. And

with respect to such things as—not so much the 35% TIM, which is sort of unavoidable—but a question of a little excessive highs and brightness is something that can be in a sense mollified or modified to an extent where it becomes more musical by reducing the high content, the tone control idea. Is there any hope for improving the musicality of recorded sound at the expense of this accuracy?

OTALA: Don't talk about musicality or any trade-offs. They've probably contaminated the word "high fidelity." Let's just take the "high" off; let's talk about fidelity. You don't need to sacrifice anything, because if you have screeching highs there, then probably something is wrong. Either you've got distortion, or you've got excessive level. Just take it off, that's it.

WILCOX: But unfortunately, a lot of it is built into the program source of the last 30 years.

RAPPAPORT: You can't do anything about that, though. If it's an equalization problem and it's simply a question of frequency response, then the tone controls come into play. But if you have 35% TIM or even 1% or whatever is above the audible threshold and it begins to sound hard and irritating, that's something you can't take away by turning down the highs or turning up the musicality control.

COTTER: Is that really true, that you can't take it away?

RAPPAPORT: You can't. Once a distortion is created, you can't take it away.

COTTER: Ah . . .

OTALA: Nonlinear distortion by its very nature is such that it is a contamination of the original signal. You can't remove it unless you remove the original signal.

HEGEMAN: Again, modulation.

RAPPAPORT: You can compensate for a distortion that you can predict. If you can predict that there is a constant frequency response imbalance in a program, you can compensate for that. If there's a constant phase imbalance vs. frequency, you can compensate for that. But you can't compensate for a distortion element that is added to the signal. A nonlinear distortion.

COTTER: There are certain kinds of time distortion, one of the cleanest examples of which would be just the vertical angle process, which puts 30% FM of the signal by itself on every stereo record ever made, approximately that for the last 15 years.

HEGEMAN: Is that why I like mono disks?

COTTER: Well, it's there in the vertical angle, is what I'm saying. It comes out when you play at the correct vertical angle and it'll vary in magnitude depending upon whether . . .

EDITOR: That's not a contamination then; it just washes out on the other side.

COTTER: No. It has a certain kind of removability. I think there is a whole family of distortions and disturbances that are separable. We ought to stop and think about that.

OTALA: As Andy says, if you can predict the distortion then you can remove it. But notably there's one form of effect that takes place which makes it impossible to predict, and that is the introduction of a frequency characteristic between the distortion generation and the distortion correction. Then you will start having a hard time to compensate for it. Especially when you start talking about networks, you would in principle be able to say, all right, we've got an amplitude response and from that we can deduct the phase response. So, all right, everything is just fine—we can measure it, we can counterbalance it, then we can redistort it, and it's okay. Unfortunately, this isn't the case, for two reasons. First of all, it is not the first pole which is important in the phase relationships—they are the second, third, and so on ad infinitum. And the second thing is, most of our transducers—right now, for instance, a few things, like cutter head, pickup, loudspeaker—are not minimum-phase networks. They are partly non-minimum-phase networks. Which means that the amplitude relationship and the phase relationship are not bound together with the Hurwitz transformation.

COTTER: They may not be Hurwitzian, and they may not even be Hilbert, but what I'm saying is, are they separable? I think that's a different question.

OTALA: Under these conditions, they are separable. If you can predict it, that means if you know the system characteristics between the distortion introduction and the distortion correction, in all dynamic domains, then you can do it. Otherwise not.

COTTER: I happen to feel a lot of these are separable distortions. They aren't simple, but I think they are separable. The acoustic recording, for instance—as a matter of fact we were talking again at lunch about acoustic recordings having novel and different kinds of time domain disturbances, relatively simple.

"... people who have been involved in these [preamplifier] tests have been using associated equipment of very, very limited resolving capability."

RAPPAPORT: The point is that the problem with this program material that we have is not necessarily its amplitude characteristics or its phase characteristics, which are separable and are predictable if you know the system, and you can correct for that kind of thing—that's relatively simple. But the problem is the time dispersive distortions that we've been talking about.

COTTER: But I'm saying those too are analyzable and in some cases . . .

RAPPAPORT: Not in all cases. Even simple distortions.

COTTER: Then that's a challenge for us.

HEGEMAN: You're gonna have a computer console, and you punch in an algorithm for every record that you have . . .

RAPPAPORT: That's right. And for every mixing console, and every microphone . . .

HEGEMAN: You get a little complicated.

COTTER: I think we ought to look to Max on this score because it's a valid thing, I think, from a music point of view to say there's an awful lot of gorgeous musical performance, from the standpoint of the artistry and the concept and the execution, that is somehow or other entrained in media that make it less than the most pleasant thing to play back in the traditional way. The question I'm asking is, are those disturbances separable so that we can recover something more than we think we have there. We know of examples; we talked about the Stockham processing of the acoustical recordings of Caruso, that's a sort of example; it can be argued that they are or they aren't. Some years ago I heard one of the retired gentlemen from Bell Labs demonstrate a very interesting processor for injecting some detail into old acoustical recordings that cut off at 4, 5 or 6 kHz by the very simple expedient of processing them with a circuit that looked for transitions and injected a little white noise in proportion to the high frequency energy. It's amazing how much crispening of . . .

HEGEMAN: Add a little noise, and the signal brightens up, I'll tell you, and you don't hear it as noise.

EDITOR: Is it perhaps for that reason that some people prefer head amps?

COTTER: Yes, possibly.

RAPPAPORT: Maybe. It's a very slight possibility.

COTTER: Anyway, the thing is that we have this tremendous wealth of recorded music. Is it all completely lost? Is it, so to speak, destroyed irrevocably?

OTALA: No.

HEGEMAN: No, I don't think so.

WILCOX: I think that we're in the kind of plastic society that we're talking about. I would personally rather devote my energies to making good recordings of the great musicians who are now alive, and think it's lucky that we do have some kind of representation of those who are either retired, or those people who have been recording for the last 80 years.

OTALA: You can easily say, all right, go to the museum and look at the old clothes. They were beautiful clothes, but you don't wear them, not them or not even their replicas. You wear *your* clothes.

COTTER: We can replicate them, though.

RAPPAPORT: We can replicate them but not with the same craftsmanship that they had, not with the same materials.

OTALA: It's a weak analogy.

WILCOX: I would like to deal with the future, not the past.

EDITOR: Max, I want to ask you a question. Of the great musicians that you've worked with—and you've worked with a number, you've worked with Rubinstein, you've worked with Solti, you've worked with some really great musicians—have any of them expressed any regret that in their early days they did not have the kind of recording techniques that they enjoy now? In other words, do they really care?

WILCOX: Some of them care. But they have again, what we were referring to at lunch as this wonderful suspension of disbelief. A musician will come in, he will be a little bit interested in the sound quality, maybe in the beginning. A few of them are very interested in it, not very many. Peter Serkin happens to be an exception; he's very interested in it. He's very interested in collaborating with me in getting a very accurate sound of the instruments. Most of them go in and immediately enter a world of fantasy, which is not an unpleasant world at all, where they're listening to the tempo and the inflection, and are not really aware how

accurately or with how much distortion their performance is being registered. I find them probably the poorest judges in some ways of recorded sound, because they don't really listen to it.
ZAYDE: But they're not looking for replication of the live instrument per se, but rather the recorded event.
WILCOX: They already know that it's not going to sound like them, and it's not that dynamic range . . .
EDITOR: They already know that it can't be done. Isn't that what it comes down to? Whereas we here know that maybe it can be done one day. This is the difference.
OTALA: Well, not really. A musician is interested in how it is played, not how it sounds. He's interested in his fellow musician, the way he articulates, for instance.
EDITOR: That's true, but I have a purpose in pressing this point. The only way to make these major record companies pursue better technology, to go in the direction that we've been discussing here, would be for these people, for Sir Georg Solti to say, "I will not record for you if this is the garbage you give me."
HEGEMAN: No, the only way to do that is to have the consumer say, "This is a lousy record" and bounce it right back at the record store. If that happens enough, that's gonna hit where it hurts.
OTALA: 80% of American records sold are pop, and there the virtue is . . .
WILCOX: 95%.
OTALA: 95%, okay. There the virtue is to introduce as much distortion as you can.
EDITOR: Actually, among the pop musicians there are more hi-fi freaks than among the classical musicians. Cat Stevens, for example, is a hi-fi nut, and I am told that his place is full of panel after panel of Magneplanars.
ZAYDE: But is he trying to recreate a sound, or a reality? We have to look at that—what is he trying to recreate?
COTTER: Or create.
ZAYDE: Right, exactly, or create. There may not be a font for developing the original experience.
WILCOX: Most pop music now is not acoustical, anyway.
EDITOR: My point is that record companies can be approached only through their pockets. If the artist is fed up with the kind of sound he's been getting, that hits the pocket. Nothing else does. The pressure comes from the technologists and the producers—I think the producer is in a somewhat better position to push than the technologist . . .
WILCOX: In this country.
EDITOR: But if the pressure comes only from that side, it's not going to be as effective as if the pressure came from the artist. Wouldn't you agree with that?
WILCOX: I think the artist is the last person it's going to come from.
EDITOR: I am afraid so, too.
WILCOX: I think it's going to come from the people who know enough about what real music sounds like, and who buy records, and say, "This is not really what it sounds like." I think there's been a refreshing return to simplicity in the last two or three years only, brought about by Doug Sax and Lincoln Mayorga, in the beginning almost . . .
COTTER: Direct to disc.
WILCOX: Who are only copying techniques of people . . .
HEGEMAN: Another reinvention of the wheel.
WILCOX: That's right, exactly.
COTTER: You had not only the fidelity question resolved in many of those old 78 recordings, but you had a kind of artistic integrity, at least for the 6 or 7 or 8 minutes that took place, that is a hard thing to find today also.
WILCOX: That's a completely separate thing. But what I see happening is that in the small record companies that are now coming along, there is an interest in something like the Blumlein technique—which I am not so interested in—but in any case, there's a return to maybe making recordings with two or three microphones. I've heard some recent recordings made with all kinds of wonderful equipment—we were discussing it at lunch time—with two or three microphones, which happen *not* to be good; all their "state of the art," all kinds of things. So you have to have someone in the control room who can actually know what an orchestra sounds like. Just reducing the number of microphones to two or three isn't automatically going to give you a terribly accurate recording of the orchestra, any more than increasing them to 36 is going to give you more clarity. But I think there's a return to simplicity in the whole thing. Certainly in my work there is, and I think I'm not a pioneer in this at all.
OTALA: I think that we have not covered, almost at all, the preamplifier. We've discussed some of the problems, but . . .
EDITOR: I'm glad you said that, Senator.
OTALA: The problem is we have circumvented the problems. There's another thing I would like to add to the agenda, if that is of general interest, and that is measurement methods.
EDITOR: Absolutely. But couldn't we follow through? I would like to get the sense of this meeting as to what some of the basic considerations are. Our subscribers are interested in components. Whereas this is also an exchange of ideas among us, and it's for the benefit of each one of us, there has to be some kind of takeout for our subscribers when it comes to what they can expect of components. So maybe we could speed up the discussion.
COTTER: We got up to the preamp, then we went off in other directions.
EDITOR: We haven't really discussed the preamp. I know there are a number of issues on which all of you differ. For example, bandwidth limiting. Again, feedback vs. no feedback in preamps.
FUTTERMAN: Bandwidth limiting is a good subject, and we haven't even touched on it.
EDITOR: We haven't touched on that at all. I would like to talk about measurements, and not only measurements but also evaluation techniques, which is of course very very close to my heart.
OTALA: One thing in particular I had in my mind when I mentioned measurement methods, for instance, is the new IHF Standard, which is just a catastrophe. It's the ultimate catastrophe. I mean we should do something about that, too.
HEGEMAN: Any time you standardize you reduce to absurdity, most of the time.
COTTER: Classes of fits and interfaces and so on, I think, no, I wouldn't agree. But I think what you're talking about is when you get a group of people to agree on a definitive method for the specification of products that they're all selling competitively, you're very likely to reach an absurdity.
HEGEMAN: Design a horse and end up with a camel.
COTTER: That's right. Can we get some opinion on what a phono preamplifier should do, in the light of all of the things we've said? Why do we have a situation where Al Foster and others feel the only difference is frequency response? Why do people take the attitude they do of specifying a group of preamplifiers which, upon measurement, verify that they all have these exquisite specifications? And yet they all sound different, and we're all striving to do better, even though the measurements are perfect. Are we dealing with time domain effects, again?
HEGEMAN: I believe so.
COTTER: Are we dealing with anything else?
RAPPAPORT: With the current evaluation of preamplifiers and the rather absurd opinion that seems to be very popular, that they all sound the same, I think there are two problems. One of the problems is the question of limits of resolution. Most, in fact all of the tests that I have heard about—I unfortunately haven't been involved in any—but that I've spoken to people about, people who have been involved in these tests, have been using associated equipment of very very limited resolving capability.
COTTER: Which we feel. You're talking about the Shure M91.
RAPPAPORT: Well, going from the Shure cartridge to the—I'm going to lose friends—the AR speakers and this kind of thing, where the kinds of distortions that they will be able to hear are *only* the amplitude distortions, because these are the only things left. We've gotten rid of, in the cartridge and the speaker, we've gotten rid of all the ability to look at time problems.
EDITOR: This raises an interesting question about testing; even though I think we should pursue the discussion of preamplifiers, this comes in at this point or at almost any point. The general question of whether garbage piled upon garbage sounds like plain garbage or a new kind of garbage. I have a feeling that it sounds just like plain garbage, which also puts a . . .
COTTER: I think what Andy's talking about is, there's a way of getting a question answered about those kinds of tests, that really reveals the nature of the problem. That is to say, when people declare there is no difference, or they talk about the only difference being that of, say, frequency response errors and when they are correct that there really are no . . .
RAPPAPORT: That's really what they heard.
COTTER: That's really what they heard, and I'm inclined to believe them. There's one question I would ask and I think I know the answer. And that is I would say, "What was what you were listening to in your comparison, something that was indeed very closely akin to live music?" And I think the answer would have to be that it was not.

RAPPAPORT: And the point is that as we get closer to live music, the differences become more and more apparent.
COTTER: In the department we're talking about here.
RAPPAPORT: In all departments. If there are x number of components in the chain, as x minus 1 of those become very very good, the last one, you'll be able to show tremendous differences in the last one.
COTTER: I'm not sure you mean what you said, because we've had some discussion in the past about this. Do you think that differences in amplitude of .05 dB become important after you've cleaned up the time domain and these other effects?
HEGEMAN: No. They don't.
RAPPAPORT: I don't think they become important in and of themselves.
COTTER: Then what you're saying is not that these things, the other things, that is more of this frequency response, the kind of thing we were dismissing a while back in connection with amplifiers, that those become important; they remain unimportant.
RAPPAPORT: They remain unimportant, but what I was saying was that as the associated components—say, we're discussing preamps—as the associated componentry becomes better in the sense of time distortions and that kind of thing, the ability of the reference system, so to speak, to resolve the differences in preamps that are time distortions or of that nature, increases. I believe the reason that these listening panels have come up with the conclusion that the only differences are frequency response differences, or cartridge loading differences—which is another very popular opinion—is that they did not have the ability to resolve the real differences in these components, which are the time dispersive types of distortions.
ZAYDE: Actually, it's even more coarse than that, if I may interject, Andy and Mitch. That is that they don't talk about frequency response, they talk about amplitude response, which is even a further coarsening of what's going on.
COTTER: But they're using pickups that introduce time dispersion, speakers that introduce time dispersion, and preamplifiers themselves introduce also these . . .
RAPPAPORT: There's a limit in the ability of the system as a whole to resolve the kinds of distortions that they really set out to listen for.
EDITOR: Does limited resolution in various parts of the system, in your opinion, invalidate straight-wire bypass tests as well?
RAPPAPORT: Yes, absolutely.
HEGEMAN: I think so.
EDITOR: In other words, if you're using, say, a Shure cartridge or some kind of low-pass filter type of cartridge, and you're using a phono stage, say, full of TIM, and then you're testing line level preamp stages by means of a straight-wire bypass test, and listening to a speaker that again has limited resolution, you don't think you'll be able to hear valid differences?
RAPPAPORT: In most cases, no.
HEGEMAN: You may well hear differences, Pete, but I don't think you can really interpret what you're hearing in terms of the differences.
FUTTERMAN: Peter, I'm an Audio Critic subscriber, and I don't know what you mean by a straight-wire bypass. Explain it.
EDITOR: It's something that others have brought up. A straight-wire bypass is when you compare the sound of a component to the sound of a straight wire.
FUTTERMAN: Wait a minute. I didn't know a straight wire has a sound.
EDITOR: But it has a signal path.
RAPPAPORT: Sometimes a straight wire has a sound too, and in many of these straight-wire bypasses that straight wire sounds very very bad.
OTALA: The pot I talked about is a very good example.
FUTTERMAN: Give me an example of a straight-wire bypass speaker.
EDITOR: Of course you can't. What about a cartridge? You can't.
COTTER: Only with electronic things.
FUTTERMAN: Even with a preamp.
OTALA: Let me phrase what annoys me the most in all those tests that have been conducted. In my opinion, basic scientific thinking is this: Everything is possible unless it has been proved to be impossible. And even if it would be proved to be impossible, then there exists a distinct possibility that the proof was wrong. Therefore, if your conclusion from a listening test is that there cannot be any differences because we didn't hear any, this is just the opposite of every

"We believe that the rise time or speed limit in the hearing perception is somewhere between 12 and 14 microseconds."

element of scientific thinking. It is the other way around. We say, in this test we did not hear any differences, therefore it is highly probable that there are differences, but we had a bad setup and we didn't find them. You have to turn it around exactly, that attitude.
COTTER: I think it goes even further in the negating direction in that I can readily conceive—and I think you would concur, Matti—that if you were to compare a pickup that has, say, a low-pass 11-kHz cutoff and even with its mechanical resonance a rise-time property that's extremely limited, with some very much faster pickup which has very much faster rise time and much greater resolution, free of needle drag distortion as opposed to the other one which, say, had a lot, free of reactance modulation, time domain shifting effects which it had none, and you were to present this signal to the input of your preamplifier under test, and compare the results obtained in the two cases, that you might well conclude that you got more junk out of the better, higher resolving system than you did with the lesser resolving system. You might infer that, therefore, the lower resolution system was "better." Now I could precisely reverse your opinion by getting rid of the problems that existed in the preamplifier, and you'd get a reverse opinion. I think that we're talking about this pain-pleasure ratio kind of thing, and the ability of the system to withstand the stress of the signals that are presented. The question behind all that is what does it take to make it sound like music?
EDITOR: That's a very good point.
OTALA: You know, there's a very funny history to this to support your thinking. A certain Professor Bougaritz, who is Belgian, has conducted a number of tests where he injected distortion into Muzak. He's a psychologist. He has recorded the level of irritation in the workers given a daily dosage of distortion in Muzak.
HEGEMAN: You don't need to add any distortion to it.
OTALA: He was really jumping when he reported that he has found a strict, direct correlation, and he was very happy. And incidentally, he has been using TIM as one of his distortion mechanisms. He was even more excited about the fact that TIM seemed to be a very irritating pollutant.
COTTER: I think the problem of getting a poor result with a better stressing signal, which is what you would want to have if you wanted to get closer to the music, is not often addressed. That's where these comparisons can go astray. Andy was talking about this with respect to the preference for level being corrected, there's amplitude of —how did Bruce put it—just amplitude variations being removed, removed all of the remaining differences. None of it was what we would call good, clean sound. On the level of stress, one of the things that I found with respect to the TIM-like processes, is that the simple introduction of a 9-microsecond rise time limitation cleaned up remarkably most anything we could encounter. We later discovered that even if we had a very ultra-clean electronics, that mechanically, accelerations—the second, third, fourth, fifth moment, whatever you want to call it—that would be implied by allowing rates of change in the principal component—we're operating strictly in the time domain independent of amplitude here—that you wind up with accelerations that are up in the magnificent G's area, and you induce nonlinearities in the mechanical systems. This flies in the face of the DC-to-light theorists, and is a subject that I think we ought to bring up at this point because the burden that is to be borne then by whatever the system is that we're considering, can't have infinite limits. Because obviously there is some stress level that will break down any system. Even if you have a gigahertz amplifier, there is some kind of signal that's going to produce an excess of garbage out with that sort of bandwidth exposure.
RAPPAPORT & OTALA: Not necessarily.
COTTER: There's an amusing little tautology, and that is you have infinite bandwidth, then the immediate reaction of connecting your loudspeaker to such a system is that it melts down because you have this infinite energy coming out of this infinite bandwidth system from the noise, just from the noise. Anyway, aside from that, it seems to me that one of the problems that we ought to avoid in an audio chain is excessive stress, stresses that are beyond the range of that which is necessary to complement the human hearing apparatus. My simple expedient seemed to work, even though that value of rise time is

clearly not discernible as different from something significantly faster or something significantly slower.
HEGEMAN: Mitch, why 9 microseconds? This is a number that I find I can't really translate into my hearing experience, or anything else. Why 9?
COTTER: We tried a lot of different values. I was looking for the longest time which would, as a final rise-time limiter in the system, not introduce something that encroached on the other limiting processes, and which still was sufficiently slower than leaving it wide open to much greater speed. I wanted something, in other words, which wouldn't be discernible as a loss in acuity; we established that that's certainly not. We believe that the rise time or speed limit in the hearing perception is somewhere between 12 and 14 microseconds. Certainly I think people would agree 12 microseconds seems to be, for an event, indiscernible from 2 microseconds. And 9 microseconds seemed to be a worthwhile limitation and yet not something that would encroach too heavily on the . . .
OTALA: Single-pole filter?
COTTER: No, not a single-pole filter. A time-domain corrected design which minimizes the maximum transition.
OTALA: A Bessel.
COTTER: Well, it's a Bessel-type transfer, it has low ringing, and it has minimum-maximum slope. A toe in other words.
OTALA: How many poles?
COTTER: Equivalent to 9 poles.
HEGEMAN: That's interesting because it's very different from my own experiments, and I admit that this is tape program, not record program.
EDITOR: That shouldn't make any difference.
COTTER: It's a time-domain synthesis, Matti, and it has less than a percent or 2 percent ripple.
HEGEMAN: We always found, and this was not only myself, the Citation II power amplifier had a 4-microsecond rise time.
COTTER: In the output, or as an input limiter?
HEGEMAN: In the output; that was what went out. At one point I designed a small amplifier for Lafayette which had a 2½-microsecond rise time, and there was a difference in the sound; now true, that amplifier sounded slightly different. But what you could seem to hear was a better airiness around the top end of the presentation. It was more real in space. It was a spatial characteristic that you heard; not necessarily a sonic thing. I've been listening for years to amplifiers that have a 1-microsecond rise time up to the full output of the thing, and frankly—I guess my ears are conditioned—I love them.
COTTER: I don't find anything inconsistent in that; it's just that I don't think the differences you're hearing come from the rise time phenomena as much as they come from other things.
OTALA: Let me put it this way. I just tried to calculate some experimental values that I've found. They boil down to some startling figures. I would say that for an amplifier having 26 dB of feedback, 0.8 microsecond is the maximum rise time that can be allowed for the amplifier itself; and about 4 microseconds is just right when you put the input filter in. But under those circumstances, and assuming perfectly abrupt slewing, 0.8 microsecond rise time seems to be right. There are some good rules of thumb.
EDITOR: But you're talking about two different things.
COTTER: But that depends very much on your definition of not only the amount of feedback but what the delay properties are in the amplifier.
OTALA: No, I'm not talking about that; that's a perfectly stable amplifier, and the delay therefore is relatively unimportant. But 0.8 microsecond comes from the simple feedback relationship. If you put a 100-kHz filter at the input, then you cannot allow the distortion to rise in the frequency range from 0 to 100 kHz. You may allow it to start rising after that, but that means open loop distortion design, to yield flat distortion spectrum.

* * *

COTTER: But this is outside of the feedback, Matti.
OTALA: I'm talking about inside the feedback.
COTTER: That's a different situation altogether. I'm talking about a rise-time limiting device that is totally passive, that is not active, that has no feedback; it's simply a passive filter. And it's the input signal that is limited to this rate of change.
OTALA: I know that.
EDITOR: Mitch, to simplify matters, could I interject something here? Your contention, Mitch, is that if in this room, or in a music room where music was being played, we could somehow preserve all the information only to a speed of 9 microseconds, and somehow got rid of the . . .
COTTER: If we'd put a gas in the air that somehow or other had a 9-microsecond rise time limitation of the form of this quasi-Gaussian, Bessel-type thing . . .
EDITOR: Then we couldn't hear any difference, you're saying.
COTTER: . . . that minimizes the maximum rate of change, at 9 microseconds slope, then I don't see that you would hear any difference whatsoever.
OTALA: I'm pointing to exactly the same thing. In my opinion, that is fully legitimate. The important difference, however, is that—I'm saying this following my own experience—say 4 microseconds would be all right as a rise time. So let's take the microphone, let's put a 4-microsecond or 9-microsecond filter after that. And then, after that I say that the amplifier following that filter must have at least 0.8 microsecond rise time.
RAPPAPORT: With 26 dB feedback. Exactly.
OTALA: With 26 dB feedback. In order to cope with the signal that is coming out from the microphone.
RAPPAPORT: So that the open-loop amplifier is not driven by the 4-microsecond rise time into nonlinearity.
COTTER: Unless it has no feedback at all, or has lesser feedback. I really don't care at all about that. I'm only saying there is some value of stress, in the sense of first, second, third, nth moment, which . . .
OTALA: Here is the big difference and the big confusion. The proponents of the so-called DC-to-light frequency response are perfectly right in exactly the fact that the amplifiers following a rise time limitation, or a frequency bandwidth limitation, must have a bandwidth to light in order to cope with the bandwidth-limited signal.
RAPPAPORT: If it's a feedback amplifier. And the bandwidth of the amplifier is determined by the amount of feedback around it. So in a situation where you have a no-feedback amplifier, the amplifier should be simply as fast as the rise time limitation, maybe a little faster . . .
OTALA: To retrace this, an amplifier having 26 dB of feedback, in order to reproduce properly a 30-kHz signal, must have a bandwidth of 1 MHz. Period.
COTTER: Yes, this is the basic Bode constraint. That's generally not adhered to, I might add, in the execution.
HEGEMAN: You noticed that, huh?
COTTER: Well, we do notice it in this interesting way. That in no system have we ever encountered less than a significant improvement in the overall sound when such a filter is used in the system between the power amplifier and the preceding stages.
RAPPAPORT: I've got a question for Matti, because you've obviously done work in this area. The question is, if you take your 0.8-microsecond rise time amplifier with 26 dB of feedback, and you put a variable rise time filter on the front end of it—and let's assume it's time-compensated and isn't going to create any problems in the audible range—do you notice a difference between 4 microseconds, 9 microseconds, or what I use, which is 14 microseconds?
OTALA: I recently conducted a series of experiments, where it was a 2-pole Bessel filter which was adjusted down from 1 MHz to the lowest frequency that I was "allowed" to try, 100 kHz. No audible effect was noted.
RAPPAPORT: And if you go from 100 kHz to say 30 kHz or 25 kHz?
OTALA: Well, I would say in a 2-pole Bessel, for instance, the problem is mostly that the amplitude characteristics are not sufficient. If you go to 50 kHz it already is ½ dB down from 20 kHz because it's so smoothly rolled off.
COTTER: You need a higher order of approximation.
OTALA: There are some other problems in higher order approximations, especially if you do it actively. So don't try that.
COTTER: Oh no, don't try it active. But passive, there are no problems . . .
EDITOR: There doesn't appear to be any disagreement among you. We all agree that the haphazard kind of bandwidth limiting will be audible for various reasons.
COTTER: There's an important amplification here. I said that I found that introducing this cleaned up a lot of the problems in the system by removing many of the TIM-like processes and some of these other time modulation effects which exist apart from feedback by itself. Matti is giving us some ideas about what a feedback amplifier would have to do that followed this situation. What we're all saying is that it's pointless to have stresses in the signal source beyond the range of that which is valuable from the standpoint

of what the hearing experience is. Are we agreed on that? We don't know what that number is precisely.
OTALA: You're saying in fact that it is pointless having an audio frequency response, or frequency characteristic, beyond x kHz.
COTTER: I prefer to talk about it in the time domain.
OTALA: Well, all right, that's your handicap.
RAPPAPORT: It just requires a little extra math.
EDITOR: Mitch, what is the relationship there?
RAPPAPORT: 0.35 over the bandwidth.
COTTER: When you leave the minimum phase situation, you enter into some other . . .
OTALA: It's 0.35 to 0.42, depending on the filter, on the type of cutoff.
COTTER: The thing is that it varies with the slope, but the essential idea is that there's some value here that we're saying doesn't remove information. But does it remove problems? I say it removes problems by preventing excessive stresses.
RAPPAPORT: It all depends on what these excessive stresses are going to do to the components. You can take an amplifier which has no problem at all with 100 kHz or 200 kHz or 1 MHz, and your filter isn't going to have an effect at all.
COTTER: What we found, Andy, that's interesting is that where you have a situation approximating that, at least insofar as we can tell, that it is the mechanical side of the system that gets into trouble. So it seems as though there's a rational basis for this, whether the electronics is vulnerable or not.
RAPPAPORT: This is a very common misconception that I've heard in conjunction with your filter in particular. I am very confident that your filter does absolutely nothing for my amplifier. But I still recommend its use in conjunction with my amplifier because most of these tweeters that people are driving with my amplifier are really giving problems.
EDITOR: That's certainly true.
COTTER: I don't disagree at all. We did have another interesting experience when discussing filters with someone, and the problem was that they were distressed to find it made a difference in the sound, because the idea of such a filter had been presented to them as something that removed a certain latent vulnerability to something, should it come along and cause a problem, rather than being an on-stream purification of the signal handling ability. Am I making myself clear? In other words, the idea behind that person's impression was that this filter was a sort of protective device—it removed lightning bolt threats, and things of this sort, but it shouldn't affect the signal. They were distressed to find that it made a difference in the sound. I had to apprise them of the fact that that was the reason we made the filter.
EDITOR: Why would you make something that makes no difference at all?
COTTER: Yes, but the point is that their attitude was that that was just a protection from those threats that might come along. My attitude was they're continuously present, and you want to remove them. I think it's important that we clarify what it is we're talking about in this connection.
OTALA: Let me read something from a paper. This is a paper published in 1970; that was the original paper on TIM. The conclusion section, if you will bear that reading, goes like this. "Conclusions: (1) To minimize transient intermodulation distortion, it is advantageous to let the preamplifier limit the frequency response of the complete amplifier. (2) In the above case, it is the power amplifier frequency response without feedback that determines the desired preamplifier frequency response. Therefore, the feedback in the power amplifier does not necessarily enhance the usable frequency response of the complete amplifier system. (3) If high-fidelity reproduction requires a 20-kHz upper cutoff frequency in the amplifier, the power amplifier should reach it without feedback. (4) In the above case, the upper cutoff frequency of the power amplifier with feedback must be at least this 20 kHz times the feedback loop gain. For instance, in an amplifier with 40 dB feedback, 2 MHz."
COTTER: Those are the classic rules.
FUTTERMAN: Yes, and my amplifier follows those rules. But I have a filter in front of it.
COTTER: Yes, which makes it immune to the range in which it could conceivably get into some problems.

"Every amplifier should have an input filter which will restrict the input signal to just that with which the amplifier has no trouble at all in dealing."

HEGEMAN: I still have a question on this 9 microsecond bit. Does that cutoff change the spatial characteristic of what you hear?
COTTER: Not so far as we can tell, Stew.
HEGEMAN: Because old Harry Olson once told me that he thought that time differences in channel-to-channel work on stereo could be down to several, like 1 or 2 microseconds, or you find yourself . . .
COTTER: The differential delay in these systems is zilch. It isn't even nanoseconds. The point is that it wouldn't matter if there was a half-second delay in a filter as long as what came out came out together, and the differential delay between the two was zilch. The differential delay even in the passband is zip because there is very minimal transient disturbance, and that's why you need a multi-order, higher order filter approximation to make the thing behave in a correct way in the time domain. The closer you move it in to the audio passband, the more significant that time domain correction becomes, because if you were to, say, have a cutoff that was really audibly inside the passband, then you might have to have your ripple component down 70 or 80 dB to be below the threshold of audibility, and that is indeed a heroic problem. So we bring it to within range; we have it outside the range of audibility; the ripple value we have certainly isn't going to cause amplifiers, or whatever, to zock from spiking or anything. And we allow the system to have some other limitations in some other portion of the system without encroaching too heavily, because these things roughly add by the square root of the sum of the squares of the rise times. So that was the feeling about 9 microseconds. Which is not a heavy feeling, not a strong feeling, so I don't mean that it's a real magic number.
OTALA: It's about 40 kHz.
COTTER: Typically around 40 kHz. That's essentially the idea.
RAPPAPORT: There's just one thing that Stew touched on, and that is that there's a very common misconception about limited bandwidth. There are those people, in fact, publishing audio reviews who maintain that unless a device has a bandwidth from DC to X-rays or whatever, or preferably below DC . . .
COTTER: That's going to collapse the space.
RAPPAPORT: That's right. The space and the air are completely gone. The idea is that this is because limited-bandwidth feedback amplifiers typically have been loaded with transient distortions, which collapse the space and take away the air and make things sound hard.
COTTER: The more feedback they have or the faster the rise time, the more time modulation effects and the more garbage there was, and the idea of going faster and faster was that you got a change in character but not a reduction in magnitude of these time modulations.
RAPPAPORT: Exactly.
COTTER: And I think that's a false goal. We've talked about that in amplifiers. But I see no reason that an audio system should be required to go beyond that. Now one of the things about digital that is a very important concept is that digital allows you to perform a sort of arbitrary approximation. You merely have to tell it, so to speak, in an abstract way, you merely have to tell the system what it is you need. You can't have infinite bandwidth; that's infinite sampling density. So you have to make a decision about what's necessary. So in a sense what we're saying here is that a 9-microsecond rise time satisfies the audio requirement, certainly. We're agreeing. Maybe we're not absolutely correct, but we're not far off the mark. You say you use 14; Stew has used 4; we're all more or less somewhere in this ballpark.
RAPPAPORT: The idea is that there is absolutely no reason for the reproduction system to be able to reproduce anything faster than that which we can hear. Unless, of course, there is the idea of a safety margin.
EDITOR: Are any of you gentlemen aware of actual experiments without electronics, in actual air, to measure this phenomenon?
COTTER: Harry Olson constructed an acoustic filter to approximate that filtering condition with some limited properties.
EDITOR: What did that consist of?
COTTER: It was a slot structure.
HEGEMAN: It was a room with a variable basic door coming down on there.
RAPPAPORT: I'm embarrassed to say I was 10 years old when I read about that in *Popular Electronics*.
EDITOR: And what were those findings?

HEGEMAN: His cutoff was basically about 7 kHz.
COTTER: 7 to 10 kHz.
HEGEMAN: 7 kHz, 10 kHz, and full range kind of thing. Everybody—and he apparently pulled people off the streets for his listening panel—but he had a fairly large sample of people . . .
COTTER: In Princeton.
HEGEMAN: Was it in Princeton? Yeah, could have been Princeton, could have been. Everybody could hear the 7-kHz cutoff. Not everybody could hear the 10-kHz cutoff, as he started, with live music, and so forth and so on. Now it's a very simple thing; you can do this in any concert hall in the world. You listen to somebody out on the stage playing a violin, and all you have to do is move forward one row from the edge of the balcony underneath it to the edge of the balcony out in the open. Your string characteristic opens up and changes absolutely, completely. But if you're under the balcony . . .
COTTER: But Stew, don't we wish we could get as good a sound as we can get just that one row back?
HEGEMAN: No! I want to be that one row front.
EDITOR: Nobody has ever tried a 35-kHz acoustical filter.
COTTER: But I'm saying if we had a sound system that would be even one row back, we'd be doing pretty good.
HEGEMAN: But not when you have it as an A-B, Mitch. That gets totally unacceptable.
OTALA: But I would make one correction on what you said, and I don't think you said it on purpose, but it might convey some misunderstanding there. You said there is no need whatsoever for an amplifying system to reproduce anything beyond that which is audible.
HEGEMAN: What is audible?
OTALA: That requires one correction. That is that in many systems, you cannot in the first place restrict yourself at the signal source to what is audible. Therefore, you cannot put the filter exactly on that point. And this goes for preamplifiers, for instance.
RAPPAPORT: What I was referring to was an idealized case.
OTALA: Exactly. So you have to design the things, the amplifiers, to be capable of coping with what's fed into them. But we're allowed to limit that.
RAPPAPORT: That's right. Every amplifier should have an input filter which will restrict the input signal to just that with which the amplifier has no trouble at all in dealing. And there's no reason for that input filter to be of any shorter rise time than that which we can hear.
OTALA: That's true, yes, fully agreed.
COTTER: What Matti is saying also, we might even have a criterion, a sort of standard for apportioning some other limitations elsewhere in the system and dividing it up say between the recording and the reproducing side, because certainly the disk cutter gets into some very serious limitations on its . . .
OTALA: And remember that there's a summation of rise times, so that your 9 microseconds and your 14 might be okay just for that case. But don't propagate that as an acceptable value . . .
COTTER: Well, the eleventh dub is going to be in trouble.
OTALA: . . . because people are starting to put ten filters in the chain, and after that you're in dead trouble, because that's 25 microseconds.
RAPPAPORT: That's right.
COTTER: Let me say this. One of the things that Stew was poking a little fun at was the use of a lot of transformers, that often happened in the past. But one of the things that was accomplished in a system that had some transformers in it, especially systems with flat amplifiers like 300B's and no feedback, was that these transformers afforded a measure of protection from these excessively fast events. Since the systems were, basically, simpler, composed of fewer links and chains, we were often dealing with systems that had—though not described in detail and in a sense almost inadvertently—very significant speed limitations in light of the sort of times, speeds and rise times we're talking about. That kind of came for free with the case at hand. Undefined, but very much a part of it. Even today, the RIAA preemphasis recording characteristic is defined as a straight line with a slope of 1 at the point at which it stops, at the top end of the audio spectrum. There's nothing to indicate that it doesn't go up infinitely, which is insane of course, because you have an infinite . . .
HEGEMAN: It's impossible to measure.
COTTER: The fact is a matter of practical significance, very little known, is that the Neumann system turns over at about 35 kHz, and that the Ortofon people originally turned over higher but modified their characteristic to coincide, because of the processing and the electronic kind of music that was being cut, to make that agree with this undefined . . .
HEGEMAN: So that's basically about a 5-microsecond turnover?
COTTER: Something like that period. You recall when Jerry Minter and you and I and RCA tried to get a cut in that characteristic?

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EDITOR: Have we got a clean signal path now, from stylus tip to power amp output? Have we neglected anything important?
HEGEMAN: I think one of the things we've neglected so far is the time adjustment of the preamp equalization network.
COTTER: Why don't you amplify on that?
HEGEMAN: As an old-timer, if you were going to measure the response of an equalizing amplifier, the simple way to do it is put in your inverse network. At which point then you can change your frequencies and go through there, and you always read the same, you return to zero on the meter, and if you have designed that inverse network properly, it's a very easy test procedure. Now you take one of these, which I have, you start putting a square wave into it, and oh boy—you have discontinuities that you can't believe. It's been my practice to do a square wave adjustment, basically a trim. Because I can't find RMA value resistors and close enough tolerance capacitors that I can afford, and I'm not even sure, if I finally came down to these esoteric values, it would always work.
FUTTERMAN: Excuse me, what's the frequency of the square wave you use?
HEGEMAN: 10 Hz to 100 kHz.
FUTTERMAN: Through an inverse network.
EDITOR: Stew, if all your stages of gain were perfectly linear, and your RIAA equalization is the exact mirror image of the RIAA preemphasis, then you should automatically get square waves.
HEGEMAN: From a practical standpoint, you either have to build a piece of measurement gear, or, if you have a production unit, you have to trim it.
COTTER: There's another problem here, and that is the RIAA boost characteristic that you use goes up how far? Where do you turn over?
HEGEMAN: I've turned it over with a 7½—since we're working at time constants of 3180, 318, 75, I took 7½.
COTTER: 7½ microseconds on the top is where you stop boosting.
HEGEMAN: That's right. It's a turnover there. Working with my existing test equipment and everything else, I could get it down to about 2½ or something like that, but then I wouldn't have the drive to drive the network to make the test.
COTTER: To interpret what you're saying, you're saying that even using a 7½-microsecond turnover on the top, stopping the boost in other words at a 20 dB up value, sloping off at 20 dB up, that you still notice that preamps go a little bananas when you hit them with that sharp a signal. Is that correct? The rise time is still going to be the rise time of whatever your generator is. The turnover is 7½ microseconds. What rise time speed do you use in a square wave?
HEGEMAN: What does my Hewlett-Packard do?
EDITOR: Probably of the order of 30, 40 nanoseconds?
HEGEMAN: It's a 10 megacycle unit; it's probably, yeah, fairly fast.
COTTER: I wonder if that's a realistic input signal in the light of what we've been talking about. 30 nanoseconds rise time isn't likely to come out of a phono pickup, not any that I know of, anyway. What would happen if you slow it up?
HEGEMAN: I don't truly know.
EDITOR: I've heard this general concern about the phase characteristics of the RIAA equalization, but one should think that whatever the phase characteristic, it would wash out with a precise inverse curve.
HEGEMAN: Peter, it's essentially a multiple network. You drive out of a low impedance. But if you're doing a passive equalization job, you do it in sections. This section takes care of this, that section takes care of that, and so forth and so on. You end up with square waves that look very queer.
COTTER: I think that there are three different problems. The first problem is that you're applying a transition that is very very fast. You said 30 nanoseconds. The second problem is that I think you're pointing to a different problem, which is that the topology of almost all the standard equalizers does not turn into a piece of algebra. If you state what the equations of that transfer function are, that resembles the transfer function stated in the RIAA or the IEC standard—simply because the components are not freed of their interaction. Even when corrections are made, as presented in a recent paper, what happens is that you may

correct the pole values but the transition has a different form of algebra, and there are saddles—there are errors in the response that develop that inevitably lead to funny-looking things in the square wave. The square wave doesn't come out looking like a square wave.
EDITOR: Yes, but he doesn't have that in his preamp.
COTTER: Well, you can do this correctly by partitioning the network so you don't have that problem. The third thing is, what do we see happening in a preamp where we don't hit it necessarily with this high a speed of transition? In other words, I would say that if you got down to the microseconds speed of transition, and you hit it with a pre-emphasized signal that's more or less like the pickup you might use—we talked about a pickup that had a 70-kHz bandwidth—you look at some of the better moving-coil pickups and you'll see a certain kind of rise time capability. What happens to a preamp if we hit it with a signal which has the 6-dB-per-octave rising velocity characteristic, coming out of a pickup such as that? Neglect for a moment the errors in equalization and talk about these time domain effects that go on. We find that most of the topologies are flawed in that they have the equalization in the feedback, and the system is capable of a lot of time modulation. The current levels at which they operate are very low f_T's, and you've got a system that's very very time-modulable by the signal. Capable, in fact, of the triggering I alluded to before that can be controlled by marginal changes in vertical angle, which is the speed of the signal transients. I think that preamps suffer categorically from a lack of understanding of what their dynamic response characteristic is. With the equalization in the system, you're boosting the sensitivity, as I've said before, to all the difference tone change that can be generated.
RAPPAPORT: If I can add something to that . . . I think what Stew is getting at is the idea that if you have a given amplitude response in your preamp, and it follows exactly the RIAA standard, and you feed it a square wave which has exactly the inverse in amplitude and phase characteristic, you should get back a square wave. But the problem there, and the reason that you can't really trim in a preamp by using a square wave—unless you know the properties of the preamp and in fact I would contend unless the preamp is properly designed—is because, especially when the equalization is accomplished in the feedback loop, you're going to get overshoots. There's going to be an overshoot all the time in that case. A lot of people are now looking at equalization and using a step with a relatively fast transition time, and they're saying well, ideally, you should get a step at the output. And that's true, but the fact that you get an overshoot in many cases doesn't necessarily mean that there's an error in the equalization. It means there are other problems occurring in the . . .
COTTER: We're getting into these other things. I'm saying that I think 30 nanoseconds is an excessive n moment, dE/dt or dV/dt or whatever . . .
HEGEMAN: What difference does it make?
OTALA: There's been some discussion about the DIM method, for instance, being invalid or too stringent in preamplifier testing. Of course, the original recommendation was that it should be inversely emphasized with an inverse RIAA curve . . .
COTTER: But you and John Curl did start to put some time limits of 3 microseconds—was it a 1 microsecond or 3 microsecond . . .
OTALA: No, no. The DIM 30 method is the normal method which has the 30-kHz roll-off.
EDITOR: That's a single-pole filter, right?
OTALA: Single-pole filter. But then for preamplifier testing we have now used the DIM 30-30 method, which has a two-pole roll-off in order to limit the rise times more or less after the inverse correction to, say, something like 30-kHz bandwidth.
EDITOR: Matti, how do you explain the fact, though, that every piece of junk that comes on the market now claims to do very well on your tests? I see this on the specs every time now. TIM, .004%; the sine-square test . . .
COTTER: He covered that in his discussion in a recent paper on these thresholds. The problem is if you average it in the right way you get some very low percentage numbers, but when you look at what that represents in noninstantaneous spaces, it turns out it's 5% or something.

"I wish that were made very clear to everyone. Do you hear us out there? Just a good TIM measurement doesn't mean you have a good amplifier."

EDITOR: All right, but they take the rms value of the . . .
OTALA: Let's put it this way. The DIM method detects primarily TIM and related phenomena. There's no problem making those vanishingly small for these bandwidths. We're still having 30-kHz bandwidth, and for that bandwidth, with present technology, it is not a problem. It was a problem five years ago when the method was originally devised. It is not anymore. There are other effects, and that's exactly why I've been here advocating a total picture, so to say. TIM is the past; it's history; it doesn't exist any more.
EDITOR: You're absolutely right. I have tested equipment here that invokes your spirit, obviously for endorsement. They say they've done your test and the device is perfect, or nearly perfect, in terms of your test—and the sound is terrible. Obviously there are other things that they haven't done right.
OTALA: That's true. There are 100 or 200 other possibilities, at least.
EDITOR: I wish that were made very clear to everyone. Do you hear us out there? Just a good TIM measurement doesn't mean you have a good amplifier! Do you all understand that out there, please?
OTALA: You can quote my words by saying that, apart from the TIM psychoacoustics, we have not done any significant work on TIM during the last five years. Everything that has come out has been done previously and has just waited in the mills of publication routine. We finished that in 1974, something like that. We considered the case closed. There was no problem any more. When it started, it was a problem, and a horrendous problem at that time. We had catastrophic amplifiers on the market in, say, 1969. Right now it's seemingly so that people have learned their lesson and they got rid of TIM. So, case closed. Let's concentrate on other factors.
COTTER: What effects do you think are still with us in preamplifiers that we should look at? Is the speed of change affecting the result? Is this rise time?
OTALA: Here we go to an area where I don't exactly know.
RAPPAPORT: You can't examine the problems of a preamp really independently of anything else, because a properly designed preamp . . . Basically you have a stage of gain, or two stages of gain, or three stages of gain, and an equalization network. The equalization network is reasonably cut-and-dried; at least it should be. Then there is a stage of gain, there is a low-level gain stage, or two, or three, or four, or five, or whatever. The idea is that the considerations for those should be exactly what the considerations are for any active stage in terms of dynamic performance.
EDITOR: In other words, the criteria we laid down in connection with power amplifiers.
RAPPAPORT: Exactly. There are a few special cases. For instance, in a passively equalized system where you have a stage of gain, then equalization, then another stage of gain, the first stage of gain has to be able to deal with some pretty fast signals, unless of course you put a filter at the input. It also has to be able to drive an equalization network which is a varying impedance, and a few other things. But it's really basically just a stage of gain.
EDITOR: How do you explain, though, the phenomenon we found here, that with a really good moving-coil cartridge aligned to the nth degree for lateral and vertical geometry, playing, say, massed sopranos singing high, nearly all preamps sound distressed—not all but nearly all. What phenomenon is that?
RAPPAPORT: It's probably the fact that there is a tremendous—because of the preemphasis and the speed of the cartridges which we're using—there is a tremendous amount of high-frequency energy. Most circuits are not designed in such a way that they will readily cope with this energ
WILCOX: Do you know that the signal is clean, Peter?
HEGEMAN: I was just going to ask Max, how do you handle that?
EDITOR: If it sounds clean and more detailed on some preamps, then the cleanliness and the resolution must be there.
WILCOX: That's the kind of signal you hardly ever find clean on any record.
RAPPAPORT: You'll find that those are the preamps which have no trouble dealing with this high-frequency information, either because of the way . . .
EDITOR: How would you quantify that kind of distortion?
COTTER: Max, I want to stop at this point and say something about that. I think the

problem that we're getting to is that we don't really know what's on the record. All we're talking about here is the playbacks that we've been making. We've seen many dimensions in which the playbacks that we've been making are nonrepresentative of what you could call the latent signal in the record. We're talking about ways of optimizing that . . .
WILCOX: I'm just talking about hearing master tapes that have distortions in sopranos.
COTTER: That's another story.
OTALA: I would like to object to Peter that the problem is—the problem can be—just the converse. That is, those that you feel are clean and nice and beautiful might just be so terribly distorted that they take all the garbage away.
RAPPAPORT: This is of course a possibility.
COTTER: That's a good kind of distortion, if it makes it sound more musical.
OTALA: It certainly is, yes.
EDITOR: I think that's most unlikely.
OTALA: Most of the distortions, when there is just slight distortion, are pleasant. Most of the distortion that I've been working with adds musicality to the sound.
FUTTERMAN: In Japan, they like amplifiers with a lot of second harmonic distortion.
EDITOR: Do you think it's possible that we have preamplifier A and preamplifier B, and we play this particular type of sound that I find particularly stressful—either solo sopranos, or massed sopranos are even better—singing well above the staff, and we use a moving-coil pickup with a line-contact stylus very carefully aligned, correct vertical tracking force, correctly adjusted anti-skating bias, everything—and on preamp A you hear a marrow-piercing scream, and on preamp B you hear a sweet, soaring and thoroughly transparent sound. Is it possible that A is accurate and B is inaccurate?
OTALA: Yes it is. I just said earlier that 90% of our subjects took the distorted sound as being the nondistorted.
EDITOR: But in the sense that I've just described there's distorted and there's distorted. I mean a marrow-piercing, screaming edge. And that's what 9 out of 10 preamps do under those circumstances. They can't take it.
OTALA: Okay, okay. If it's really that bad, then okay. But if it's slightly less, then there's a possibility.
COTTER: Matti, there is one way of working with a system where you think you may be on one side or the other of a threshold effect, and that is to iterate the system. We talked briefly about this before. If you think you are below threshold, and in fact you are actually above threshold, if you go through several such units, inverting and playing again and inverting and playing again, and you can't hear any difference between n such things strung in series and one or none, then isn't it a fair surmise that you are still subthreshold?
OTALA: Well, possibly.
EDITOR: That was a pretty heavy sentence there. You're suggesting a whole system of testing there.
OTALA: I would suggest another system of testing, specifically for The Audio Critic.
COTTER: Without knowing what the details are, why—I'm not saying you'll find out why this way—you'll just find out whether or not you're above threshold.
OTALA: One way of testing would be to make a distortion box that could generate different kinds of distortions, and the no-distortion position would be absolutely clean. You would simply try, with the kind of situation you described, introducing more distortion. Now if one gets worse and the other one gets better, then you know which way things go. That, I think, is psychoacoustically the only possible way of detecting distortion.
FUTTERMAN: You might be cancelling distortions.
OTALA: Even so, yes.
EDITOR: You have to know what you start with. And this is one of the problems, because you never know exactly what you start with.
OTALA: Well, you can iterate there too.
COTTER: You're saying that you really ought to standardize it. I'm saying that this kind of approach allows you to make a determination based on a belief that if you've got some process of this sort, then if you iterate you're going to increase the magnitude of what you've got. Now it's hard to imagine processes that are purely multiplicative that are not going to work something like that. It seems to me that it allows you to tell whether or not there's a change, and therefore whether or not you're near or below the threshold. But it doesn't tell you what the mechanism is. As you aptly mentioned before, that you can synthetically introduce a certain value of distortion where you can get a good correlation of discrimination, but the jury, though consistent, is unable to tell you the character of what it is they're tracking in making the identification. When you're dealing with near-threshold phenomena, this is possibly not an unlikely occurrence. I think what Peter is talking about is something that's a more gross difference, and I think it relates somewhat to this effect that the vertical angle changes induce sharp changes in some cases and much less sharp changes in others. We seem to be able to connect those with trigger-like—call it TIM or time shift—effects that a lot of people have described . . .
EDITOR: This is not TIM in Matti's classic sense.
COTTER: No, but this is a time area thing. And it is not unusual to find that sort of thing described by observers who have some experience as mistracking. Because it has the sound of a fast, high-frequency mistracking. We were surprised to find that it was not mistracking at all, but it was electronic in character. I think there remain problems in the speed time modulation processes within preamps that go beyond equalization.
OTALA: Let me also state one thing which is in parallel with your thinking. The DIM test, as it normally goes, only detects amplitude variations, more or less, of the 15-kHz signal.
COTTER: Unless you look at the sidebands we talked about.
OTALA: Yes. We are presently thinking in terms of—if that proves to be feasible—trying out a method where the same test signal setup is used but possible phase modulation effects on the 15 kHz carrier are detected. That should in principle provide a more sensitive measurement method.
COTTER: Do a direct frequency detection, then look at the output of the frequency detector?
OTALA: Yes, with a phase-locked loop. That would probably be first of all a simpler test, and secondly would relate directly to the time domain effects in such a way that a fast-rising slope of, say, a 3-kHz signal would then phase modulate, in the time domain, the 15-kHz signal component.
COTTER: You have now got a heroic filter problem in your instrumentation, though, to eliminate your signal components from your detector.
OTALA: No, not necessarily. The phase-locked loop technique is quite okay for that purpose. That *is* a heroic filter, by the way.
COTTER: I'm saying it's a Hilbert filter, but it's still a speed problem.
EDITOR: Where do we stand here? Are we more or less squared away on the signal path from stylus tip to power amplifier terminals? Is there anything significant to be added, anything this group would like to contribute to . . .?
COTTER: We said you should pay attention to all of these time domain effects; we didn't tell how. And we all, I think, disdain the classic THD and even some of the TIM measurements, because they're not sensitive to these things.
EDITOR: Le't talk for just a little while about this multiple pass test that you suggested. Is this your recommendation in lieu of the straight-wire bypass test with its obvious shortcomings?
COTTER: I think that if you think you have something that's subthreshold, that you're in a much better position to believe that it is subthreshold if you can put some significant number of those elements in series, and cannot track the difference between n elements and 1 element or no element. I've found very few pieces of equipment that will do that. I've never found, other than some of the experimental things we've built and the new thing we've made, two preamp stages that can be strung in series without distress, without an obvious . . .
EDITOR: Exactly how do you do this? You pad out the first one to reduce the . . .
COTTER: Pad it out, inverse equalize it back again. Very simple. Very straightforward. We don't use 7½ microseconds; we use a 35 kHz turnover. Close to the same.
EDITOR: Is it conceivable that inaccuracies introduced in the course of the first pass will mask all further deterioration?
COTTER: You ultimately become limited in your n, in your choice of n, by heroic accuracy problems. Even using the best possible components, and you start to get your networks turned out . . .
OTALA: With the far-out poles.
COTTER: Not only the poles, but the fact is that you've got the accumulating error. If you do five passes, and you're talking about a quarter of a dB being a JND, you suddenly require less than .05 dB in everything. And that's getting hairy.
EDITOR: Let's just take one device, one pass versus two passes.
COTTER: I think one versus two, if you can get it to that stage, you are outside the realm of most everything else.

EDITOR: But we're back to this question that I asked quite a while ago. Does garbage overlaid with garbage sound different from plain garbage?
HEGEMAN: I think it sounds like more garbage.
OTALA: More garbage is more garbage, and that's it.
EDITOR: But does more garbage sound different from plain garbage?
HEGEMAN: If you have an A-B chance to listen . . .
EDITOR: I'm not arguing, I'm just asking.
COTTER: Not always. For instance, we know from the work on flutter that the mind reaches a saturation value of flutter at about 1.4% rms at a 3-Hz flutter rate, such that you cannot distinguish as greater or lesser 5%, 2%, 3%, 7%. They are indistinguishable in quantity, in quantitative value. So that there is for the time shift effect, at least in the low speed rate area, a kind of saturation effect. There's another set of experiments that deal with flutter rates in the area of 30, 35 Hz modulation rate that also show a kind of saturation effect. If you accept a place theory, and a place theory spread kind of critical band idea, for pitch and flutter sideband recognition, then obviously there should be some kind of saturation value for the time shift, more of which is not going to be discerned as more.
EDITOR: In other words, two passes through a really bad device may not sound different from a single pass through that bad device.
COTTER: You've got to go back and ask that other question we asked. Is this sensibly like music, to start with? If Max would walk out of the room, then maybe the comparison isn't valid to start with. But if Max is intrigued by the musical sound—and we're taking you for argument's sake as a sort of paragon of musical judgment—if you would accept it as decently musical, then I think you ought to be concerned with whether it sounds different. If it's horrible to start with, then obviously I think that the comparison . . .
EDITOR: Mitch, why is this preferable to padding the preequalized stage out to unity gain and then comparing it against a straight wire?
COTTER: But how do you get a straight wire out of the pickup that you can listen to?
EDITOR: You can't get it straight out of the pickup.
COTTER: I submit that a pickup signal has all kinds of things in it that are very different from what you get out of, say, a tape recorder synthesis.
EDITOR: That's true. There's much more out-of-band energy, for one thing.
COTTER: That's right. I think you have to be able to handle that realistic signal. Having it work well on a tape recorder synthesis doesn't necessarily prove that it's going to be as good or even better than something else, when presented with . . .
RAPPAPORT: I was just going to say that Mitch's test, the iteration test, has one thing in common with the bypass test. That is the limit of resolution. As you say, if you start out with something that's decent, something that's good, and you're able to construct a padding network, or for that matter a straight-wire network that's valid, that has no sound, that has no adverse loading characteristics or anything like that, then the test is valid. However, if you have a situation where you have a gross limit in resolution or you have some kind of distortion that's occurring, then there is the possibility that something's going to be masked. I think it's safe to say that if two trips through a device sound worse than one trip through a device, something is wrong. Which does not necessarily mean that if it doesn't sound different, something is right.
COTTER: I think, though, that with the experience we've had, you don't have to be quite as wary and look for quite as far-out things as you might suspect or you might want to consider. In the abstract it sounds that way but as a practical matter, when we've done these things, most of these systems emerge as grossly faulty where bypass 1 and bypass 2 are . . .
RAPPAPORT: There's a possibility that the gross fault may be due to your interface. It's not that cut-and-dried.
COTTER: We try to control those things, and I just think that in the case of phono preamplifiers, the difference is a lot bigger a difference than has been suspected. When we wind up with an improved system that's been designed this way and we listen to it, the differences are readily apparent differences, and they are in the direction of improved clarity, improved musicality. So I think what we're finding is that we've been looking with our traditional methods at aspects of the preamplification or low-level signal processing that don't consider these time shift effects.
RAPPAPORT: All I'm saying, though, in response to this test and also the straight-wire test, because they are very similar, is that if the interface has a fault, if the interface is audible, and that's taken as being a transparent given, then any test that utilizes that interface is going to be faulty. I think that most of the straight-wire bypass tests that are being performed right now are using a faulty interface. And sure—the straight wire is going to sound different than even a very good component, and a bad component is going to sound the same as the straight wire. It's the same thing with the iteration test to a certain degree, because through one trip, you don't hear the interface. Through two trips, you hear the interface. And if there's a problem in the interface, then two trips are going to sound different than one trip.
EDITOR: There is a difference, though.
COTTER: You have to contrive to make your interface relatively inert.
RAPPAPORT: You have to make sure that it is absolutely inert, not relatively inert.
COTTER: The problem is that nobody can say for certain that a certain amount of reactance change isn't a factor.
RAPPAPORT: That's the problem.
COTTER: I think again we're digging far deeper than you need, because there are certain kinds of things that come to light. One of those areas is a very great difference in the time modulating properties of systems that pass even an imperfect interface system, as contrasted with systems that just are grossly faulty, where you get a one-two comparison that's just a black and white scene. You don't have to stop and think or do any extended switching.
RAPPAPORT: The point is that those cases are audible without an iteration test or without a straight wire test.
COTTER: Then you're relying wholly on judgment. This is just a way of keeping you honest.
ZAYDE: What you're saying, you should buffer the straight wire to render interfaces as invisible as possible.
RAPPAPORT: The point is that once you begin to buffer you're assuming the transparency of the buffer.
ZAYDE: That's true, too. You're damned if you do and damned if you don't.
EDITOR: This is the superiority of the multiple pass test, that at least your first interface is typical of in-use interfacing.
RAPPAPORT: But that's only valid for one trip. Two trips, you begin to see an abnormal situation, and the iteration test is only of use if you begin with two, three, four, five, six trips.
COTTER: If a network that is a very high impedance, and whose reactance is a small part of the kind of load that a preamp is working into, is sufficiently disturbing to affect it, then perhaps there's a problem also. What I'm saying is that I think you're overkilling small problems when in fact if you do this, it turns out there's some grossly faulty . . .
RAPPAPORT: I have one fear, and that is that ultimately I think this discussion is going to appear in print. And your iteration test on the surface seems like a fantastic thing, and . . .
COTTER: Don't fly into it with the feeling that it's very easy to do. One can spend weeks building inverse equalizers that don't affect things.
RAPPAPORT: I've told you before that I've performed this test; in fact, we discussed this a few weeks ago. The idea is that I would hate to see a rash of tests like this that involve op-amp buffers and all kinds of things, and people are saying well, three trips sound the same as one trip.
COTTER: Horrors. No question, Andy, a very good point.
EDITOR: As a matter of fact, somebody once sent me a bypass box with an op-amp buffer in it. And, indeed, the device that it was supposed to prove to be transparent was indistinguishable from a straight wire.
COTTER: So was the box.
RAPPAPORT: I don't want you to think that I'm condemning Mitch's test, because in a very real sense I stole it from him. But you have to be very careful.
COTTER: But it's tough. And you work on components, because it carries us back exactly to the idea that Matti introduced, with just the pot. The pot can be a distorter in a very peculiar way. Capacitors are very very big problems, especially . . .
EDITOR: At this point we're sort of picking at the remains of the subject on the plate. It might be time to pass on to loudspeakers.

* * *

***Editor's Note:** The rest is all about loudspeakers, with only minor digressions. Our reasons for not publishing this portion of the transcript were explained above, in our brief introduction to Part II. Thus our two-part series ends here, not with a bang but an outtake.*

Speaker Wires and Audio Cables: Separating the Sense from the Nonsense

"There's the full moon, Dr. Van Helsing. It's time to connect the pure silver cable." And other such widely debated matters.

There was a time, not so very long ago, when speakers were connected to the amplifier with lamp cord stripped at the ends and wrapped around screw terminals, and the other components in the system were plugged in by means of plain, off-the-shelf phono cable. Those innocent, hype-free days are gone forever; specialized, high-technology wiring has become *de rigueur* for the serious audiophile, in much the same way as a jogging suit for the jogger.

Is this just another trendy affectation or does it have some basis in electrical science? Are the differences real and audible, or are they wishfully mythicized by cultists? Some time ago we set out to find the answers by measuring the relevant electrical characteristics of a large number of wires and cables of different configurations and listening to each in a known reference system. Initially we even considered the possibility of a brand-by-brand test report, such as we might publish on speakers or preamplifiers, but we soon realized that our results would then be subject to simplistic misinterpretation by those who can't live without numerical scores and rankings. A systems approach that takes into consideration the basic nature of electrical interfaces quickly dispels any notion that one particular brand of wire or cable can end up as the "winner" even if it happens to have certain advantages over others under given circumstances. It simply isn't the same kind of problem as finding out which preamp is best. We shall therefore concentrate here on the overriding issue of reality vs. fantasy, with only incidental brand recommendations.

Copper, silver, platinum or kryptonite?

One superstition should be disposed of right up front, before we get involved in more complicated matters. Please note once and for all that electrons retain no memory of the metal they have flowed through, be it copper, silver, gold, platinum or whatever. This, of course, has nothing to do with the use of silver or gold to reduce small-area contact resistance and oxidation, which is a totally different subject. But you may rest assured that an electrical signal that has traveled through a length of silver cable is absolutely indistinguishable from what it would have been if it had traveled through a length of copper cable of equal resistance and reactance. Anyone who tells you the contrary is either an outright charlatan or a duped victim of vampire tales about the nature of metals. We have checked this out with some very advanced students of the periodic table, metallurgy, solid-state physics and electromagnetism, and they just turn their eyes heavenward with a God-give-me-patience expression when confronted with the silver cable fad in audio. We have also performed some fairly conclusive experiments of our own. To wit:

We set up a simple, unambiguous but very high-quality single-amped stereo signal path by completely turning off the tweeters of a pair of Vandersteen Model II speakers, substituting a pair of Pyramid Model T-1 ribbon tweeters (properly phased and pulse-aligned with the midrange), connecting this speaker setup to a perfect sample of the Hafler DH-200 power amp (which happens to be very happy with this load), and driving the whole thing out of our Reference A "front end" (mostly Cotter). Our assumption was that any changes made in the line-level cable between the front end and back end of this system, if indeed such changes altered the signal quality, would be easily audible and unequivocally attributable to the cable differences. To magnify the possible differences, we used 10 meters (32.8 feet) of cable in each case.

The big shoot-out was between 10 meters of **MLAS**

Ltd pure silver coaxial cable and 10 meters of **M.A. Cotter Co.** triaxial cable (similar to the former **Verion** triaxial), with identical top-quality phono plugs soldered to each. Although pure copper is only 5 percent less conductive than pure silver, there was a greater difference between the two cables because of the gauges and constructions involved; the Cotter inserted 7.2 ohms in series with each channel, the MLAS only 0.73 ohms. The shunt capacitance per channel added by the full length of each cable was 900 pF with the Cotter and 800 pF with the MLAS. None of these four numbers had the slightest significance at the interfacing impedances of the hookup. More significant was the fact that each cable had excellent RF shielding; the triaxial could be expected to be somewhat superior in this respect under extreme conditions of RF interference, but the important point is that the comparison was not between a well-shielded and a poorly shielded cable, as we suspect had been the case in some of the listening tests cited by the silver cultists. RFI can definitely cause audible degradations of the signal, and some audio cables are very marginally RF shielded. But that has nothing to do with silver vs. copper.

Need we spell out the results at this point? Blind testing by a number of exceptionally keen-eared auditioners revealed, in the words of the French chef in that notorious TV commercial, "no differawnce!" Both cables sounded exactly the same in repeated A-B substitutions. If there was the slightest preference for one of the cables, it was perhaps in favor of the copper triaxial, but ever so rarely, vaguely and inconsistently. (Do we have more RFI in our lab than we think?) We call it an obvious draw. And we call silver cable, at $24 per meter per channel (without plugs!) an exploitation of the moneyed audio neurotic, precisely the sort of thing that makes high-end audio appear fatuously snobbish and repellently doctrinaire in the eyes of so many intelligent but nontechno-freak music lovers.

It shouldn't really be necessary to add that the "100% pure copper" fad, which is the equivalent syndrome in speaker wire selection (heavy-gauge silver speaker wire being too costly even for faddists), must be classified in the same subdivision of vampire lore. Because another thing that electrons can't remember, children, is whether or not the length of metal they have flowed through contained a few little impurities. The only possible electrical effect of the latter would be a minute change in resistance. We haven't performed any experiments in this area, since we don't have a metallurgical laboratory; nor do those who profess to hear an improvement when some manufacturer tells them that this here is 100% pure copper wire, yes sir.

Then why does it sound better?

This brings us to the broader issue of why perfectly levelheaded and open-minded audiophiles hear an improvement when they replace their old audio cable or speaker wire with a new super-fidelity design. We're inclined to believe that most of the time the perceived difference in sound is really there—what they report is true but not necessarily because of the superior quality or technological advantages of the new wiring. Here are some of the possible alternative explanations:

Breaking an old metal-to-metal connection that has been undisturbed for many months and may be oxidized to some degree could in some cases significantly reduce contact resistance, diode effects (i.e. rectification) and the resultant nonlinearities, some of which may have been marginally audible. This is somewhat more likely at the higher impedances and lower signal levels of front-end electronics connected with audio cables than at either end of the speaker wire. In other words, just the self-cleaning action of breaking the connections and plugging back the old cables may also make everything sound better.

Or consider this. In an RF-infested environment such as most of us inhabit, where every cubic foot of space is teeming with CB, police radio, TV and innumerable other signals, the entire cable and wire harness of a home stereo system acts as a huge "antenna farm." Adequate grounding and shielding, combined with proper circuit design, will minimize RFI in the front-end electronics, but many power amplifiers are also vulnerable to, and largely unprotected against, RFI backing into their output terminals via the speaker wire. Since RF antennas are tunable and directional, just moving the speaker wire around can change RFI sensitivity for the better or worse, let alone changing the construction of the wire, which can also have an effect but with little or no predictability. (A possible exception: **Mogami** speaker wire, which is of self-shielding coaxial construction and may conceivably provide a consistent minimum level of RFI protection, although that's not what Mogami believers confess as the prime article of their faith.) Thus the improvement or deterioration heard in the sound after changing the speaker wire may be simply an RF antenna tuning and orientation effect.

Another cause of audible differences, almost invariably but quite incorrectly attributed to the inherent virtues or deficiencies of speaker wires, is the effect of small series inductances and of small shunt capacitances on the stability of certain amplifiers. Any typical length of speaker wire represents a series L, ranging from a fraction of one microhenry to 8 or 10 or more microhenries, and a shunt C, ranging from a few dozen to a good many thousand picofarads. (This in addition to the series R everyone talks about in connection with power loss and damping factor.) Those values are right in the ball park for either stabilizing or destabilizing a feedback amplifier that can be upset by complex load impedances. A perfect example is the Bedini Model 25/25 power amplifier reviewed elsewhere in this issue. With a 2-microfarad capacitor plugged directly into its output terminals while passing a low-amplitude square wave into an 8-ohm resistive load, this amplifier goes into uncontrolled oscillation and blows its power supply fuses. Nevertheless, when driving electrostatic loudspeakers that are fairly closely modeled by such an RC load, the little Bedini is happy as a lark and sounds gorgeous. Why? Because the couple of microhenries of series inductance introduced by the speaker wire provides a stabilizing trim that isolates the speaker capacitance at the higher frequencies. (The amplifier appears to have no such protection internally.) Now what if we switched to some ultralow-inductance speaker of the braided variety (**Polk**, etc.)? The isolation would be greatly reduced, a marginal oscillatory condition might reappear, and innocent golden ears would conclude that the new speaker leads are of poor design because the sound is now worse. Whereas if the amplifier had been, say, the Rappaport AMP-1 (now defunct, alas), which is almost totally insensitive to load capacitance, there would have been "no differawnce." Conversely, the lower series inductance and higher shunt capacitance of the braided cable might have been a highly beneficial trim for some other slightly peculiar amplifier circuit loaded by some other slightly peculiar speaker. There are more things in amplifiers and speakers, Horatio, than are dreamt of in your cable philosophy . . .

Other facts and other fictions.

The idea that a length of cable conducting an audio signal can be modeled as a transmission line, with a characteristic impedance (like 300-ohm TV antenna lead-in), is another fallacy that needs to be disposed of here. Transmission-line effects can begin to come into play when the length of the transmission path is at least a quarter wavelength of the frequency being transmitted. For argument's sake, we'll call 40 kHz to be the highest frequency of interest in the accurate reproduction of music. (See also Part II of the seminar transcript, elsewhere in this issue.) The electrical quarter wavelength of 40 kHz is well over one mile. The electrical quarter wavelength of 18 Hz, at the bottom end of the useful audio spectrum, is over 2500 miles. Therefore, to talk about transmission lines and characteristic impedances in audio wiring is the rankest nonsense. There are no signal losses at audio frequencies due to mismatched cable impedances. If your Polk Audio Soundcable, for example, sounds better than the conventional speaker wire you were using before, the reason is *not* that its characteristic impedance (the square root of its L/C ratio) has been carefully manipulated to come out at 8 ohms, to match the nominal impedance of your speaker. You already know what some of the real reasons might be.

Then there is the much-discussed *skin effect*, the tendency of alternating current to flow near the surface of a conductor rather than uniformly through its entire thickness. This is claimed to create too much resistance to signals at the higher frequencies unless very finely stranded wire is used, with each strand individually insulated (i. e. litz wire). Again, an examination of the basic physics of this phenomenon shows that it becomes significant only as we approach the megahertz region. We have measured the 10 kHz AC resistance (that is to say the R component of the total Z at 10 kHz) of many different types of speaker wire and can report that it exceeds the 1 kHz AC resistance by no more than 5 or 6 percent in worst-case examples. Since the entire speaker wire represents only 1 to 4 percent of the total load in a typical speaker installation, the power loss due to skin effect alone is most likely of the order of one or two millibels (1mB = 0.01 dB) at the higher audio frequencies, totally swamped by the more significant though still small losses due to series inductance. Once again, academic bugaboos without a quantitative perspective lead only to equipment hypochondria instead of audible realities.

As for those inductive losses, let's look at a worst-case possibility. In the latest modification of the Beveridge 'System 3' speaker, the impedance of the highly capacitive electrostatic "line source" is approximately 1 ohm at 20 kHz. Pretty hairy. Now let's assume that in a biamped setup the line source is driven through 15 meters (just about 50 feet) of ordinary No. 14 speaker wire. (Would any audiophile use thinner wire for such a long run?) We measured 0.66 microhenry per meter in this type of wire, so that the total series inductance comes to 10 microhenries. That represents an impedance of about 1¼ ohms at 20 kHz. Combined with the resistance of approximately 0.3 ohm of the 10-meter wire, the total rms impedance at 20 kHz comes to just a little over 1¼ ohms, let us say 1.3 ohms. In other words, there will be a larger voltage drop across the wire at this frequency than across the speaker itself, reducing the 20 kHz voltage drive to the speaker by something like 5 dB compared to a noninductive connection of the same resistance. Now *that's* not academic; it will cause a significant roll-off that should be avoided, especially since the speaker happens to be already rolled off to some degree in the top octave. The solution would be to place the amplifier near the speaker and use as little wire as possible, or alternately to switch to a very low-inductance speaker cable like the Polk and hope it doesn't make the amplifier go unstable with the unisolated capacitive load. This is obviously an extreme example, chosen to prove that speaker wire inductance can be an issue, although more often than not you can safely forget about it. What you shouldn't forget is that the "best" choice can turn into the worst under exceptional circumstances. Understanding the total system is the only insurance.

The realistic criteria.

What, then, are the genuinely desirable characteristics of audio cable and speaker wire as we step out of the Transylvanian night vapors into the broad daylight of scientific inquiry?

In the case of shielded cables going into and out of the front end of an audio system, we believe the most important criteria are good, clean contacts and effective shielding against both hum pickup and RFI. The ultimate solution, covering all bases, would be a triaxial cable with Camac connectors. Unfortunately, there exists no audio equipment today ready to accept such a cable; in fact the Verion/Cotter type of triaxial cable with conventional phono plugs is still somewhat difficult to interface with equipment having coaxial jacks grounded on the shield side. As for Camac plugs, only Mark Levinson equipment accepts them and only the two-wired kind. In any event, avoid audio cables with cheap, flimsy plugs and light open-mesh or single-spiral shielding. And never trust a connection until you have tugged at it and found it unshakable and totally noise-free.

A good dielectric, such as Teflon, is also an important requirement in a quality audio cable; dielectric materials chosen with cheap and easy fabrication in mind often exhibit capacitance changes with varying signal frequency and voltage, which may in extreme cases be the cause of spurious modulations of the signal. Since the dielectric is seldom specified in ready-made audio cables, price is generally the best indication of quality, although there may be unfortunate exceptions.

Very low capacitance per unit length matters only in audio cables driven from a high-impedance source. For example, the excellent little Precision Fidelity C7 tube preamplifier, which has no flat-gain line amplifier stage, may present an output impedance as high as 7000 ohms on account of the level potentiometer used in its passive output section. This preamp definitely needs low-capacitance output cables, especially for longer runs, otherwise the high frequencies will be rolled off. On the other hand, a preamp like the Hegeman (Hapi), with its 15-ohm output impedance, couldn't care less about the output cable capacitance. You could use 50 meters of the highest capacitance cable you can find and it wouldn't make a bit of difference. Or take tone arm cables. With low-output moving-coil cartridges the cable capacitance is immaterial; with moving-magnet and moving-iron cartridges it enters very much into the correct load calculation. Again, it's the difference in impedance. Don't let anyone tell you that you *must* have low-capacitance cables for good sound; ask him to explain why that would help you at the impedances that exist in your particular system.

In speaker wires, the situation is also far from black-and-white. Low DC resistance is generally mentioned as the most important criterion, to minimize the influence on damping factor and to waste as little of the available amplifier power as possible. Granted—but, again, watch out for quantitative reality as against qualitative theorizing. The DC resistance of **Monster Cable**, advertised as "the high-definition speaker wire" and almost as thick as a pair of pencils, is a very impressive 0.01 ohm per meter (0.003 ohm per foot). Conventional No. 14 speaker wire is only half as good in this respect: 0.02 ohm per meter. But a 5-meter length of No. 14 and a 10-meter length of Monster Cable would each add exactly 0.1 ohm to the source impedance seen by the speaker and/or the load impedance seen by amplifier, so that you can't talk about a "better" wire without specifying how much you're using. If you have the amplifier close to the speakers and need only 2 meters (6½ feet) of speaker lead per channel, we solemnly guarantee that you won't hear a difference even if you use No. 18 wire. On the other hand, for wiring a 15-meter long recreation room with the speaker leads routed around the baseboard, Monster Cable would be highly desirable. You've got to think numbers, not labels.

Series inductance can also be critical, as we've seen, but it seldom is—and when it is, you must know whether you need higher or lower inductance for your particular amplifier/speaker interface. Most wires you would normally consider for a quality installation are in the range of 0.5 to 1 microhenry per meter; only the Polk type of braided speaker cable is of a totally different order, measuring as low as 0.035 microhenry per meter. With typical amplifiers and typical dynamic speakers exhibiting a rising impedance at the higher frequencies, you need not worry about this criterion.

The capacitance of speaker wire should be of no consequence to an amplifier that can drive electrostatic speakers with their incomparably higher capacitance, but a few amplifiers can be made unstable specifically with medium-capacitance loads in the double-oh to single-oh microfarad region and should therefore not be connected to the speakers with the Polk type of cable, which is 10 to 40 times more capacitive than others. The speaker wire with the lowest capacitance measured in our tests was the imported **ILV Lucas**, closely followed by ordinary No. 14. Anything in the 40 to 70 picofarads-per-meter range can be considered very low-capacitance speaker wire.

The most farfetched idea about speaker wire performance comes to us from France. It calls attention to the possibility that the distance between the plus and minus leads will be minutely varied by the magnetic field force between the two wires as well as by the acoustical energy in the listening room. This would cause a fluctuation of the energy storage in the speaker leads and thereby modulate the audio signal. Wild, isn't it—but not completely without plausibility, especially at current levels of several amperes, which are quite common in loud playback through large amplifiers. According to this theory, very rigid speaker cable with solid (unstranded) wires will minimize the effect. We have absolutely no opinion on the subject but are willing to concede that this kind of undesirable modulation might be marginally audible under worst-case conditions (such as the third round of Pernod without water). We haven't been able to verify it.

In fact, we've been able to hear very few and only very small differences among speaker wires and audio cables in our tests so far—and none that couldn't be easily explained by one or more of the considerations discussed above. But then the moon may not have been in the right phase, and unfortunately we were fresh out of wolfsbane . . .

Recommendations

Since there are obviously no unqualified "bests" in this product category, we just want to mention informally some brands that have given us good results.

For the most effective hum and RFI shielding, nothing we know of equals the now extinct **Verion Triaxial** audio cable. The **Cotter** company (Verion's successor) is using almost exactly the same cable at the output of their transformers and electronic modules but has just barely begun to make it available as a separately purchasable product. Their version, incidentally, has a greatly improved phono plug with a springy ground contact that always grips tight but slips on and off with ease. Really nice.

In a very low-capacitance shielded audio cable, our favorite so far is **Denon Audio Cord** (of the order of 50 pF per meter). It's thick, rugged, very limp, and comes with high-quality phono plugs.

Our favorite deluxe speaker wire at the moment is **Monster Cable**. At 0.01 ohm, 70 pF and 0.7 microhenry per meter, it seems highly suitable for just about any application except long, long runs to crazily capacitive speakers such as the Beveridge System 3. Also, it's very limp and flexible, with a transparent vinyl jacket that allows you to inspect the condition of the finely stranded wire at any point. What's more, Monster Cable dealers are equipped to prepare the ends of any length of Monster Cable exactly the way you want it, with spade lugs, Pomona-type double banana plugs, color coding, etc. A very classy, well-thought-out product.

Our second choice would probably be **ILV Lucas** cable, which is somewhat higher in DC resistance and series inductance but quite a bit lower in capacitance. It may be more readily available to our overseas readers.

As for ordinary speaker wire, No. 14 or even No. 16, there's not a thing wrong with it. Unless there's a special problem as discussed above, you're unlikely to gain anything by switching from it to one of the super cables. No. 18 should be used only for very short runs.

In all cases, remember—electrons obey only the laws of nature, not the dictates of fashion in the audio salons nor the incantations of the high-end shamans and warlocks.

A Spate of Speaker Systems, Large and Small, Good and Bad

By the Staff of
The Audio Critic

This is a motley group indeed, including some fairly impressive new developments and some major disappointments, names that you're unlikely to have encountered and names that have some cachet, with little or no relationship discernible between price and sonic accuracy.

By now we ought to be used to the idea that most speakers are pretty bad, regardless of price, and that the next one we test is statistically likely to be bad, too—but we just can't think that way. We still start every new evaluation with naively high hopes, only to end up being astonished once again that hardly any speaker designer appears to have a reference standard for reasonably accurate sound. We firmly believe that a side-by-side listening comparison with one of the few speaker systems capable of a fair imitation of music, such as for example the 25-year old (!) Quad electrostatic, would instantly send most of the new designs we examine back to the drawing board. Obviously, no such comparison takes place, the designer relying instead on some highly deficient or wishful private criterion.

Another source of constant astonishment in our laboratory is the obvious lack of a comprehensive test program in the vast majority of loudspeaker factories. The dozen or so simple, straightforward electronic and electroacoustic tests we listed in our cumulative reference issue (Vol. 1, No. 6, pp. 14-15), to which we routinely subject every speaker we review, reveal in nearly every instance vulgar design errors that typically would have cost no money at all to avoid or correct. What are these manufacturers measuring?

Interestingly enough, the "raw" drivers most audiophile-type speaker houses buy from the limited number of trade sources (such as Audax, Becker, KEF, Peerless, Philips, etc.) are quite good; it's the way the complete systems are conceived and executed that shows a consistent avoidance of homework. Filter theory, in particular, seems to be foreign territory to all but a few speaker designers, and their crossover networks are lamentably primitive as a result. We hate to keep harping on this, but it happens to be perfectly true that the level of engineering competence currently accepted as normal in the hi-fi speaker business would get anyone fired in two weeks at a good aerospace firm. That's because aerospace vehicles with bad electronics kill people; bad speakers only kill the joy of music and only for the finicky listener.

Beware of speaker fuses.

The one new thing we want to add this time to our previously established principles in the area of speaker design is a warning about speaker fuses. Too few designers realize that only a heavier fuse (say, from 5 amperes up) can be considered as a straight-wire segment of the speaker leads. Lighter fuses, especially very small ones of the order of ¾, 1 or 1½ amperes fast blow, constitute highly nonlinear, current-variant circuit elements that can audibly modulate the signal. This is quite a different problem from the relatively high resistance of such fuses, which has long been of concern to audio purists because of its effect on damping factor and power dissipation. Here we're talking about a phenomenon that actually introduces veiling, edginess and other amplifier-type sonic crud. (See, for example, the Axiom and Magneplanar reviews below.)

The question to ask, then, is whether or not you absolutely need all that protection for your oppressively fused speaker. Are you driving it with a huge amplifier? Do you ever drop the stylus with the volume control wide open? Do you play a lot of electronic synthesizer music with loud, continuous-wave signals in it? Is the speaker really so fragile? If the answer is no on all counts, you'd be better off shorting out the fuse or at least replacing it with a somewhat heavier slow blow or a much heavier fast blow. Also, check the speaker fuses that come incorporated in some amplifiers as part of the package. The signal obviously doesn't care at what point in its path the fuse sockets are located, be it at the amplifier or speaker end of the wire; what matters is the type of fuse and its value.

The only permanent solution will be a more sophisticated protective device that doesn't depend on temperature and melting to create an instant open circuit and at the same time looks like a straight wire to the amplifier.

Axiom TLT-1

Axiom Engineering Laboratories, 9601 Owensmouth Avenue, Chatsworth, CA 91311. TLT-1 Loudspeaker, $500 the pair. Five-year warranty. Tested #100279 and #100280, followed by #100433 and #100441, on loan from manufacturer.

First the good news. This 38-inch tall floor-standing column speaker is very well constructed, considering its price; the exceptionally heavy particle-board structure incorporates a diagonal dividing wall that doesn't quite reach the bottom, thereby creating a 6-foot folded back-wave path for the 8-inch woofer/midrange, ending in a vent to the rear of the column. The 1-inch soft-dome tweeter is, *mirabile dictu*, in phase with the cone driver (though they are far from pulse-aligned, either geometrically or electrically), and the resulting sound has surprisingly good definition and focus, probably accounting for the speaker's enthusiastic following in some circles. The bass is quite solid, though the tuning of the enclosure evidences some anomalies; the −3 dB point of the system is somewhere around 48 Hz.

Now for the bad news. We find the upper midrange and lower treble much too hard and edgy, often to the point of nastiness. We couldn't live with this kind of persistent irritation for any length of time. Initially we thought there may be something wrong with one or both of the tweeters; the manufacturer then sent us a second pair of speakers, this time with Ferrofluid injected into the domes. The revised TLT-1's showed peaks, troughs and mild ringing located at totally different frequencies than similar flaws in the first pair, but these were all within the range of the 8-inch driver; the basic edginess of the system above that range remained the same.

Since the Peerless KO-10 dome tweeter used by Axiom is known to us to be quite capable of producing smooth and musical highs (although Axiom puts a pinhole through the dome for wrongheaded reasons not worth discussing here), we concluded that the main cause of the ugly cutting edge in the region where the tweeter takes over is a miscalculated crossover. The tweeter cuts in at 2 kHz, which is much too low for a tiny 1-inch dome, and its drive level is set much louder than that of the 8-inch unit. (No tweeter control is provided.) Thus the tweeter tends to protest when the strings dig in or the soprano hits a high note—in other words, when there's a lot of energy near the bottom of its overextended operating range.

Another problem is that Axiom puts a tiny fuse in series with the tweeter: ¾ ampere in our original pair of speakers, 1 ampere in the more recent pair. (See our comments on such fuses above.) When we substituted a much heavier fuse, the hardness and edginess were somewhat reduced but not eliminated, alas. Too bad, since the TLT-1 is a decently priced, audiophile-oriented product such as the mass-marketers would never dream of making, but that's still not sufficient reason for us to recommend it at its present stage of development.

Beveridge System 3

Harold Beveridge, Inc., 505 East Montecito Street, PO Box 40256, Santa Barbara, CA 93103. System 3: unamplified electrostatic line-source speaker with integral woofers, $3500 the pair. Unlimited warranty on parts; five-year warranty on labor. Tested early production samples, on loan from manufacturer.

Here it is—the Beveridge electrostatic system *without* built-in electronics that owners of expensive power amplifiers and electronic crossovers have been waiting for. In some ways it's the best loudspeaker we've ever tested, in some ways it's woefully deficient, but luckily the deficiencies are in large part curable and, we're told, in the process of being cured.

The heart of System 3 is still the 6-foot Beveridge electrostatic "line source" with its ingenious cluster of acoustical waveguides. The electrostatic elements as well as the acoustical lens have been somewhat reworked since System 2SW; conventional transformer coupling to your choice of amplifier has replaced the integral direct-drive tube amplifier; the bottom of the line source's working range has been raised to approximately 200 Hz, below which two 10-inch woofers per side take over. The latter are now incorporated in the same very handsome 6½-foot high tambour cabinet as the line source; the cylindrical cabinet is split lengthwise, the back part forming the sealed enclosure for the woofers, one on top radiating upward, the other on the bottom radiating downward through a slot. A built-in 200-Hz passive crossover with 18-dB-per-octave slopes can be bypassed for biamped operation. Thus the new Beveridge (which doesn't replace its predecessor but provides an alternative to it) is a monolithic full-range electrostatic/dynamic hybrid system.

The net result is a substantial improvement in dynamic headroom over the 2SW, essentially the same uniquely persuasive radiation characteristics and spatial perspective, a few minor new problems in the line source, and completely unacceptable bass—at least in our very early production samples.

Let's dispose of the bad news first: the entire cylindrical cabinet is *alive*, its surface acting as an uncontrolled passive radiator that partially swamps the woofer output and may also (we aren't equally sure about this) interfere with the line-source output. Thumping the cabinet makes it go "waoom?" with a questioning inflection. Our measuring microphone showed a tremendous amount of spurious energy right off the wall of the enclosure; at 152 Hz there was a very loud and buzzy high-Q resonance; furthermore, the 10-inch drivers themselves exhibited various anomalies that overlapped the cabinet resonances, making valid measurements rather difficult. For example, the electrically measured system resonance appeared to be at 33 Hz with a Q of 0.7, but the acoustically determined dynamic Q (in response to a step function) appeared to be much, much higher, in the region of 1.5 to 2.0 at low level and somewhat beyond 2 at high level. The −3 dB point *may* have been at 33 Hz as predicted by theory, but too many other things were going on in the bass, including a rather sharp breakpoint or "elbow" at 26 Hz a few dB further down and various peaks and suckouts all over the place. Rather messy, all in all, and the sound confirmed the measurements; we simply couldn't duplicate or even come close to the solidity and definition of our Janis W-1 reference woofers no matter where or how we set up the Beveridge.

Harold Beveridge has informed us that all of these bass problems are being taken care of. The cabinet has received more effective bracing and deadening (one of our spies thumped the latest version just before we sent this review to the typesetter and reported that the "wa-oom?" was gone); the drivers are also being revised; our major objections should be past history by the time we do our follow-up report for the next issue. Or so we hope.

Moving on to more pleasant observations, we still consider the radiation geometry of the Beveridge line source—the way it launches the wave front into the room—to be the most nearly correct of any speaker system developed so far and, combined with the inherent time-domain advantages of an electrostatic transducer, capable of the most successful imitation of a live sound field. As soon as we heard the first few bars of a choral recording through the System 3, we knew that we could never go back to the Koss Model One/A, not even as the midrange of our triamped reference system. There was just no comparison; the spatial characteristics were so much more distinct, three-dimensional and lifelike. Many music lovers will forgive the System 3 (or any other Beveridge model, for that matter) all its sins because of this one overwhelmingly persuasive quality. In comparison with the remembered sound of System 2SW, we gloried in the new freedom from strain at really loud levels and the absence of any built-in amplifier peculiarities (we used the Rappaport AMP-1), but we also noted some unmistakable midrange colorations well above the woofer passband as well as a very slight softening of focus. As we said, the liveness of the cabinet may have spilled over into the working range of the line source, which is after all embedded in the same structure; in addition, we measured some new anomalies. Among the latter were bad ringing on tone bursts at 942 Hz and 1.28 kHz, and square-pulse response that was not quite as astonishingly flawless as in the 2SW, though still better than that of dynamic speakers. We must reserve final judgment on this pending the promised cleanup of all mechanical resonances.

The frequency response of the line source can also be expected to be different in more recent production samples than what we measured. Beveridge supplies a passive equalizer network as part of System 3, to be inserted between the preamp and power amp or into the tape monitor loop. The network synthesizes a broad peak around 10 kHz to compensate for a drop in the unequalized response and also provides a few dB of variable boost at lower frequencies. It became obvious, however, that part of the reason for the declining frequency response of the electrostatic unit was the interaction of the latter's capacitance with the speaker wire's series inductance (see the full story in the wire and cable article elsewhere in this issue). This effect was further aggravated by the 1-ohm resistor Beveridge had wired in series with the transformer primary in early samples, with a view to limiting the amplifier-draining impedance drop at the highest frequencies. The resistor has been removed in recent production (the resulting impedance at 20 kHz is now only 1 ohm!); the equalizer network characteristics were under review when we last inquired; the whole frequency response situation is still up in the air at this writing. One thing is certain: there wasn't much to be heard or even measured above 12 kHz in our early sample, and that surely needs to be fixed before the speaker can be considered for reference applications.

You're beginning to get the picture: the early samples should never have been released. That's par for the course, however, among small manufacturers of exotic audio components. The first ten or twenty users are always the guinea pigs. As consumerists, we see red every time that happens, but the Beveridge is such a special product that we're willing to be a little more patient. We know we'll never be able to live for any prolonged period without that unique wave launch characteristic and the resulting sonic illumination of the room. In all other speakers the sound appears to dribble out of little holes, at least by comparison. So we're keeping the faith until we receive our revised samples; meanwhile, if you happen to be one of the early birds who just couldn't wait and bought the first pair available, we suggest you get in touch with the factory and discuss the possibility of a retrofit. In our opinion, they owe it to you.

B&W DM7

B&W Loudspeakers Ltd., West Sussex, England; distributed in North America by Anglo American Audio Company Inc., 1080 Bellamy Road North, Scarborough, Ont., Canada M1H 1H2, and PO Box 653, Buffalo, NY 14240. DM7 speaker system, $1190 the pair ($1350 the pair in special finishes). Tested #08813 and #08814, on loan from distributor.

Striking in appearance and impressively low in sonic colorations, this English import would be a strong contender for our "Reference B" selection were it not for the rather steep U.S. price, which reflects the extra middlemen involved in an export-import situation. In England it costs less, but in any country it would have to be considered a beautifully made and quite excellent-sounding speaker.

Not much larger than a very large bookshelf speaker, the DM7 stands on a columnar pedestal to reach a height of three feet, its bullet-shaped dome tweeter sitting externally on top of the cabinet, covered by a hemispherical wire basket. Not exactly conventional. The rectangular enclosure houses an 8-inch bass/midrange driver (with an unusual plastic cone that appears to be woven like wickerwork) and a passive radiator of approximately the same size. The crossover between woofer and tweeter is rather elaborate, with some obvious attention to phase relationships and delay compensation, although the tweeter is connected with reversed polarity relative to the woofer, so that absolute coherence is impossible by definition. Nevertheless, pulses are quite accurately reproduced (a la DCM and Tangent, with a little opposite-going preshoot); pulse shape retention is good to 0.14 msec width (only a very few speakers make it to 0.10 msec or less). Most remarkable of all, tone bursts elicit no ringing whatsoever at any frequency. That's truly rare and may account for the basically neutral tonality of the DM7.

The −3 dB amplitude response corners of the speaker are at 50 Hz and 17.4 kHz, and everything in between is spectacularly smooth. On-axis tweeter response is down only

10 dB at 40 kHz; up to 13 kHz the tweeter radiation pattern is close to spherical. Really nice. The fourth-order Butterworth alignment of the vented enclosure is pretty much in the ball park; the Q of the system appears typical for this alignment but migrates a little with increasing drive. All in all, we can't find anything very serious to criticize in the basic design of the DM7 except that out-of-phase tweeter connection; of course, it's still a fairly small two-way dynamic speaker, with all the inherent limitations of the breed.

The crisply defined, spacious, transparent and largely uncolored sound of the DM7 is equaled or excelled, among all the dynamic speakers we've tested so far, only by a last-minute arrival, the new Vandersteen Model IIA (but not by the Model II). The one thing that might place the DM7 lower than the IIA in our sonic ranking is just a touch of nasality, which is pitched higher and is therefore less noticeable than the typical midrange ''nah-nah'' of most speakers. The considerably lower price of the IIA clinches our preference. The DM7 is sufficiently impressive, in any event, to make us look forward to testing B&W's all-out Model 801 with a great deal of anticipation.

DCM 'Time Window'
(further improved)

DCM Corporation, 670 Airport Boulevard, Ann Arbor, MI 48104. 'Time Window' floor-standing loudspeaker, $660 the pair. Five-year warranty. Tested #12676 and #12677, on loan from manufacturer.

DCM has replaced the Philips woofer/midrange drivers in the Time Window with superior units of in-house design, has made some minor changes in the vented system alignment to correct the tracking of the vents with the woofers, and is now individually matching the Philips dome tweeters to each speaker. The result is greatly improved overall sound, considerably less colored by the characteristic lower-midrange thickness of earlier Time Windows and more clearly etched as well. The dynamic headroom, already very good from the start, has also benefited; with a good amplifier the improved version is capable of reasonably clean sound pressure levels that will crack the plaster in most listening rooms. If you're an SPL freak, this is the medium-priced speaker for you, without a doubt, unless you're willing to consider low-fidelity alternatives.

The amplitude response ''corners'' of the Time Window are now at 44 Hz and approximately 16 kHz, which isn't half bad for a not very large system of fairly high efficiency. What's more, the speaker is quite smooth and free of any obvious ringing between those frequencies; square pulses retain their shape down to a width of 0.15 msec but show the inevitable opposite-going tweeter preshoot that goes with polarity reversal. (We've had endless friendly arguments with designer Steve Eberbach about this.)

We observed no serious misbehavior by the speaker in any of our tests; the woofer voice coils do come out of the gap, of course, when you drive them extremely hard, at which point the Q goes to pieces, but there appears to be adequate control up to quite a high level. (Ultimately there's no substitute for a big, grown-up woofer, and DCM has come out with one, called the Timebase, which we plan to review in the next issue.)

If it hadn't been for the last-minute arrival of the Vandersteen Model IIA, this latest version of the Time Window would have been our unequivocal ''Reference B'' selection, since it surpasses the Model II by about the same margin as the latter bettered the older Time Window. The IIA, however, is even more neutral, focused and generally accurate overall, with an airier and more three-dimensional rendering of space, although the new Time Window is also very good in all these respects, as well as more efficient and capable of playing considerably louder without breaking up. Our choice is the Vandersteen, but we know that some users will be happier with the DCM, especially in view of its significantly lower price. For the money, we don't know of any speaker that even approaches the Time Window in transparency, inner detail, spatial perspective, naturalness and just plain musicality. It's one of the few engineered products among a hundred amateurish cut-and-try boxes competing for the same consumer dollars.

Fried Model C
(improved)

Fried Products Co., 7616 City Line Avenue, Philadelphia, PA 19151. Model C satellite monitor, $950 the pair ($400 the pair in kit form—everything but the wood). Tested samples on loan from manufacturer.

Fried has made a few minor changes in the Model C satellite, which have resulted in major improvements in both measurable and audible performance. This is a very good little speaker now, one of the better (though not the best and certainly not the best per dollar) dynamic systems around. We still hear a slight edginess and some midrange coloration, but nothing to complain about vigorously; the overall sound is open, detailed and highly listenable in the speaker's working range, which is from 75 Hz on up. A subwoofer is, of course, mandatory.

It amuses us that *one* of the two irrational and ineffectual ½-inch holes we originally protested against has been plugged up, bringing down the dynamically observable Q (in response to a step function) from about 2.0 to 1.1 in sealed-box terms. That's much better damping, but plugging up the one remaining hole makes it better yet! Maybe in the next ''improved'' version . . .

The pulse response is even better than before; we observed very good pulse shape retention down to 0.1 msec width. Tone bursts appear to cause no ringing to speak of, anywhere in the audio range. There still seems to be a peak in the amplitude response, at about 2.5 kHz and measurable only with some difficulty but easily apparent to the ear when the speaker is swept. This may explain the slight cutting edge on music. The culprit is the quasi-horn-loaded dome tweeter, which we never particularly liked, although Fried swears by it on account of its power-handling ability.

Incidentally, we listened again to (but didn't measure) a pair of the very similar Model B/2's, modified exactly as the Model C's. This time the smaller and cheaper B/2 didn't sound better than the C, proving that the C had indeed been meaningfully improved. The B/2 sounded fairly neutral, but from time to time an annoying upper-midrange hardness intruded, and the openness of the C was in no way equaled. The B/2 needs, and benefits from, a subwoofer even more than the C; it opens up quite a bit when its small woofer isn't driven below 100 Hz. We still prefer, however, the Vandersteen Model IIA as well as the latest DCM Time Window to any Fried satellite speaker, with or without a subwoofer.

Magneplanar Model MG-I

Magnepan, Inc., 1645 Ninth Street, White Bear Lake, MN 55110. Magneplanar Model MG-I speaker system, $495 the pair. Tested #129680, on loan from dealer.

This junior version (one 5-foot panel per side) of the "Maggie" distributed-voice-coil dipole design had been advertised to us long before we tested it as the most natural-sounding, musical speaker anywhere near its price, far preferable to our favored Vandersteens, DCM Time Windows, etc. We fully appreciate those aspects of the MG-I listening experience that incite such partisanship; what we don't quite understand is why these partisans can't hear the obvious flaws that make the MG-I an inaccurate and, on balance, unacceptable full-range speaker.

The qualities that seduce the MG-I fans, whether they know it or not, are the relatively large-area launch of the wave front, the upper-midrange smoothness and the excellent coherence (we measured accurate pulse shape replication down to a width of 0.12 msec). We like all that, too; music in the real world doesn't come out of two or three different little holes, with a different delay through each, and it helps when speaker design is cognizant of that reality. The MG-I does indeed present a rather natural sonic perspective and a focused image, especially when you reduce the rearward radiation with a blanket or overcoat. (Permanent padding to attenuate the back wave, as in the Quad and Sound-Lab electrostats, would of course be the better solution.) But there's more to a good full-range speaker than just that—and the MG-1 hasn't got it.

For one thing, the speaker doesn't stop speaking when the input signal stops. (See also our more detailed discussion of the inherent energy storage problems of the Magneplanar design in Vol. 1, No. 6, pp. 19-20, under the Tympani I-D review.) Tone bursts excite easily measurable ringing in the MG-I throughout its range; at 3.04 kHz and also at 8.7 kHz the situation gets completely out of hand, resulting in some of the worst ring patterns we've ever observed in our laboratory. This explains the piercing and ultimately very fatiguing quality of the MG-I's highs. We certainly couldn't live with that very long. A step function causes totally uncontrolled ringing in the bass panel; if this were a sealed system we would guesstimate the equivalent dynamic Q to be somewhere between 10 and 20—that's how long those ripples keep coming. Luckily, the amplitude response of the speaker rolls off very rapidly below 60 Hz, so that all low-frequency anomalies come through heavily attenuated. (This is known as "tight bass" in some circles, meaning that only the accentuating harmonics of the bass notes are audible and the hard-to-handle, rumbling, high-energy fundamentals are missing.) On the top end the speaker is equally deficient; above 10 kHz the response drops sharply and becomes nonexistent past 18 kHz. All those apparent highs are actually lower-treble ringing. Thus the most appropriate use of the MG-I would be as a midrange driver between, say, 100 Hz and 2 kHz, a range in which its best features dominate.

By the way, the sound of the speaker improves quite noticeably when its very light protective fuse is shorted out. (See our discussion of speaker fuses in the introduction to these reviews.) On the other hand, the manufacturer threatens to void the warranty should any tweeter trouble develop as a consequence of such tampering. In any event, the aggressive highs are still there without a fuse, but they burn the ear a little less and the overall sound is somewhat more transparent.

None of the above should be interpreted as denying the obvious first-time appeal of the MG-I when you switch to it from a conventional medium-priced dynamic speaker system of conventional radiation characteristics. To our ears, however, the appeal is of short duration, and our instruments provide ample evidence to back up our ears.

Onkyo Model F-5000

Onkyo U.S.A. Corp., 42-07 20th Avenue, Long Island City, NY 11105. Model F-5000 Phase Aligned Array Speaker System, $999.90 the pair. Tested #09072742 and #09072746, on loan from manufacturer.

It was the four-color ads that originally attracted our attention to the three-way, floor-standing Onkyo F-5000. They were saying all the right things, about the importance of preserving the "specific phase *(time)* relationships" of the original signal, about "a *total* solution to the phase problem," about laser interferometry and computer analysis as design tools, about "electrostatic-like clarity, definition and center imagery," and everything else that's dear to our heart. Just imagine—a marketing-oriented Japanese mass-producer with a truly sophisticated, up-to-the-minute speaker design philosophy! It was too good to be true. And, in fact, it wasn't true.

When a positive-going pulse is applied to the input terminals of the F-5000, the woofer moves forward, whereas the midrange and the tweeter move backward. How any manufacturer can talk about the decisive importance of phase accuracy in a speaker system and then not drive all voice coils with the same polarity is completely beyond us. The trademarked "Phase Aligned Array" simply isn't phase aligned! It's quite incapable of a genuine replica of a pulse. There are also serious energy storage problems in the highly publicized planar (flat-diaphragm) drivers; the tweeter and the midrange liberate energy for a period three times as long as the pulse duration. This is time-domain integrity? The woofer is a little better in this respect, but on tone bursts it rings at many frequencies in its upper range; the midrange exhibits a particularly nasty ring at 2.035 kHz; and even the unusual "direct-drive" polyamide membrane tweeter, which amazed us with its flat-to-40-kHz range, rings at 13.2 kHz. This tweeter also does something we've never experienced before; when fed a sine wave at no more than a reasonably high power level, it synthesizes *half* that frequency—not a higher harmonic, but f/2! Would you call that a breakup or a breakdown of the diaphragm? All we can tell you is that it sounds really weird.

Where the "phase aligned" Onkyo shines is, paradoxically, in good old garden-variety amplitude response. We'd call it essentially flat and very smooth from 40 Hz, which is its −3 dB low-frequency elbow, all the way to 40 kHz. At 50 kHz it's down only 10 dB! The damping of the sealed-box woofer is a bit on the loose side, our step-function test indicating a dynamic Q in the neighborhood of 1.5, reasonably independent of drive level.

The sound of the F-5000 is a tremendous letdown after all the ballyhoo. Music comes out hard, raspy and edgy, especially on female voices (it could be that 2 kHz ring); speech is hollow and garbled. Spatial focus is simply nonexistent, a sure indication of time-domain trouble. "Center imagery" indeed! Let this be a lesson to those who question our emphasis on the products of small specialist companies; someone like Richard Vandersteen or Dr. Roger West could have told Onkyo in two seconds what was wrong with their sound.

Perspective MK2

Laboratoire d'Etudes et de Developpements Holophoniques, Paris, France; distributed by MBR Electronique, 15 r. Remusat, 75016 Paris. Perspective Model MK2 floor-standing loudspeaker, $2750 the pair (U.S.A. price). Tested samples on loan from distributor.

This almost four-foot high floor-standing column speaker, known as the Audience I in a slightly different earlier incarnation, is something of a cult item in France, where claims of towering and self-evident superiority to all other speakers are made about it with some regularity. That happens to be sheer fantasy; however, the main reason for this review, which is based on mid-1979 samples and could with some home-stretch haste have made our last issue, is that the Perspective is no longer just an esoteric Gallic curio. It has meanwhile become a full-fledged U.S. of A. rip-off, being offered in stateside high-end salons for $2750, which is an insult to the intelligence of the American consumer. There's very little more material and labor in the Perspective MK2 than in the Vandersteen Model IIA at one third the price. The cabinet of the Perspective is heavy and well made, it's true, but four drivers from the Audax catalog and a first-order crossover network couldn't possibly bring up the price to that staggering level, even with shipping, import duty and fat profits for all middlemen.

The design goal of the Perspective is the commendable one of time-domain integrity; as far as the alignment of the drivers is concerned, this is basically achieved, with everything in phase and quite good pulse replication (though not nearly as perfect as illustrated in the Perspective literature, which doesn't show the trailing ripples of energy release after pulse cutoff that we observed). The rather eccentric and complicated woofer enclosure design doesn't particularly impress us; using a second woofer internally as a kind of compression and rarefaction valve for the air load behind the outside woofer is wasteful of amplifier power and doesn't really accomplish anything that couldn't be duplicated by any number of less devious approaches. The system doesn't know and doesn't care how its Q and its f_3 are synthesized, and there's certainly no difference in sound quality between two systems of the same Q and f_3, no matter how differently these were achieved, as long as the air-moving capabilities are the same. So the simplest way, not the most original, is obviously the best way.

In the case of the Perspective, the f_3 (the −3 dB frequency) is 30 Hz, which is rather impressive, but harmonic distortion ranges from 9% to 12% between 25 and 60 Hz at not unreasonable power levels, indicating that perhaps the woofer design priorities leave something to be desired. As far as amplitude response is concerned, the entire system is very flat from 30 Hz to 30 kHz; we have no complaints on that count. The moment of truth, however, comes when the speaker is tested with tone bursts. About the Perspective you don't ask at what frequencies it rings. You ask, at what frequencies *doesn't* it ring? And the answer is—don't ask. It's just awful. At every imaginable frequency we measured profound ringing. You have to hunt for a point in the audio spectrum where the tone-burst response improves to bare decency. In addition, at 760 Hz the woofer runs into a self-cancellation problem, accompanied by a particularly strange kind of ring.

The net sonic result of this is rather unpleasant. The speaker sounds shrill, aggressive and unfocused, so much so that we could hardly believe that our samples weren't defective. But if they were, how come both measured and sounded the same? One would expect defects to be randomized.

Both the designer and the distributor of the Perspective MK2 have expressed to us the opinion that Americans don't know how to listen to music or musical reproduction critically. If they're right, the speaker has a fair chance of succeeding in the U.S. market.

QLN I

QLN Audio, Eklandagatan 6, Box 5061, 402 22 Goteborg, Sweden; distributed in the U.S.A. by Scandinavian Sounds, 233 Esplanade, San Clemente, CA 92672. QLN I two-way minimonitor, U.S. price NA. Tested preproduction samples, on loan from manufacturer.

A Swedish attempt to beat Rogers, Fried and all the others at the satellite/minimonitor game, the QLN I partly succeeds and may entirely succeed yet, as we have word at press time that both the bass/midrange driver and the crossover network have been improved since our tests.

The design consists of the KEF T27 dome tweeter, a Dalesford cone driver for the bass and midrange, a rather complex high-order crossover network with time-delay compensation, and a surprisingly heavy and rigid little sealed box, only slightly larger than that of the Rogers LS3/5A. The woofer Q is approximately 0.6 at all drive levels, indicating superb damping; the -3 dB point on the bottom end is 80 Hz; the frequency response is extremely flat and remarkably smooth all the way up to 35 kHz; tone bursts provoke no bad behavior worth discussing, at any frequency. So far so good.

Now for the bad news. The tweeter is wired out of phase with the cone driver, so that absolute time coherence becomes an impossibility. Nevertheless, the speaker reproduces pulses pretty accurately, with the typical opposite-going tweeter preshoot also observable in the similarly flawed DCM Time Window, Tangent RS2, B&W DM7 and others. A worse flaw is that the crossover design brings in the KEF T27 tweeter at too low a frequency for comfort. We measured full passband level out of the little 3/4-inch (19 mm) button at 3.5 kHz; that's zapping it a bit too hard in our opinion. An even greater problem is that the crossover network appears to present much too difficult a load to nearly all amplifiers. The same amplifiers that handle immensely taxing loads like the Beveridge and Sound-Lab electrostats without batting an eyelash go into some degree of oscillation when driving the QLN I. We didn't bother to run a vector impedance curve on the speaker, but there must be a mind-blowing capacitance across its terminals.

The end result of all this a a fairly accurate, transparent and focused sound, with none of the midrange thickness of so many dynamic speakers, but rather hard, bright and edgy, as well as slightly sibilant. After a while it becomes fatiguing. The overdriven tweeter and the ringing amplifier are the most probable culprits; we're hoping against hope that the announced ''minor change in the crossover'' will correct these two problems, if not the tweeter polarity. The QLN I could then be a potential winner.

Sound-Lab R-1
(formerly SL-6000)

Sound-Lab Electronics, 5226 South 300 West, Suite 2, Salt Lake City, UT 84107. 'Renaissance Series' R-1 electrostatic loudspeaker, $2200 the pair (without woofers). Tested #0058 and #0059, on loan from manufacturer.

Now this one is a real contribution to the loudspeaker art. Even in its present, early production form, we prefer it by some margin to the similarly early Beveridge System 3 we tested, except for the latter's unique wave-front launch (i.e., radiation geometry). Needless to say, both designs can be expected to undergo refinement as the months go by, but at this writing the Sound-Lab electrostatic unit is considerably more neutral in sound and also measurably freer of assorted anomalies in response. The Koss Model One/A, which we never liked on the extreme top and bottom, must also take a back seat to the Sound-Lab in midrange neutrality and transparency. What's more, the Sound-Lab can play louder than any other electrostatic design we know of.

The technical brain behind the new speaker is Dr. Roger A. West, who worked for many years with Arthur Janszen back in the days of the KLH Nine and always wanted to make a better electrostatic utilizing all the advanced ideas accumulated along the way. The SL-6000, now rechristened the Renaissance Series R-1 (huh?) upon the advice of a marketing man, is the end result. The design is crossoverless from approximately 100 Hz on up; below that you can use any subwoofer you wish. Sound-Lab supplies on request an entirely conventional subwoofer for which they make no special claim; we used our trusty Janis W-1 with Interphase 1 electronics. The electrostatic transducer consists of five separate vertical strips forming a curved panel approximately four feet high by two feet wide. A rather attractive latticework covers the panel and divides it into five times seven little rectangles, a grid of 35. A snap-on cushion fills out the concave rear of the panel to attenuate the back wave and reduce typical dipole anomalies to a minimum. (Cf. our comments on the Koss One/A in the context of our ''Reference A'' system in the last issue and on the Magneplanar MG-I in this issue.) The system is transformer coupled but presents a somewhat higher impedance to the amplifier across most of the audio spectrum than is optimum loading for typical transistor circuits, making up for the power loss by very high efficiency. The advantage is that even at the highest frequencies the impedance doesn't quite drop to the quasi-short-circuit values of the Beveridge and other highly capacitive speakers. (See the Beveridge System 3 review above.)

That transformer is responsible for the most obvious idiosyncrasy of the speaker, a sharp peak in amplitude response at 23 kHz, apparently caused by some kind of LC interaction between the transformer secondary and the transducer element. With the misnamed ''roll-off frequency control'' of the speaker wide open (meaning all the way up to the 33 kHz position), this peak is at its maximum of 9 dB. Playing around with the control, you can bring down the peak at the cost of creating a big dip just below it. The optimum setting—not much peak, not much dip—seems to be at the 25 kHz position. So it's a resonance damping trimmer, not a roll-off control.

This and other possible reactive peculiarities of the transformer may account for the unsettling effect the speaker has on many amplifiers. If an amplifier has the slightest tendency to go hard and edgy on program material with lots of upper midrange and lower treble energy, the Sound-Lab R-1 will bring it out. On the other hand, Class A or near-Class-A amplifiers like the Bedini Model 25/25 and the JVC M-7050 remain sweet and smooth right up to clipping when driving the Sound-Lab. A pair of Amber Series 70's strapped for mono, though not Class A, would be another good match, as they hardly ever clip while remaining almost as sweet and smooth. We also noticed some strange wrinkles in the otherwise excellent pulse profile, centered approximately 0.16 msec from the leading edge and roughly corresponding to the half wavelength of 3278 Hz, which we determined to be a ring frequency in our tone burst tests. Whether this is a transformer problem or an energy storage and delay condition in the transducer, we're not sure; nor do we know whether it has anything to do with whatever amplifiers may find indigestible in the speaker. It may take us a while to sort out the complexities of this not exactly basic cookbook design. It isn't flawless, for sure.

Otherwise the Sound-Lab is rather impressive in measurable performance. From about 150 Hz on up to 21 kHz, the head-on amplitude response is almost dead flat; the low-frequency elbow is at 120 Hz and a rather large resonance at 145 Hz, so that a somewhat higher crossover than 100 Hz would be desirable, but there is adequate output still at 100 Hz. Dr. West tells us that the bottom-end resonance of the system is subject to some judicious design manipulation and that this problem can be eliminated. The segmented geometry of the active surface is almost surely responsible for the lobes we observed in the radiation pattern; however, there was no serious ringing synthesized by the interference patterns, the one frequency discussed above being the only genuine ring point as far as we could tell. The lobeyness of radiation necessitates extremely careful speaker placement; about 7 or 8 feet apart and turned 45° in toward each other seems to be the position that minimizes the problem to the point where it isn't at all obtrusive.

Despite this mixed bag of test results, the total sonic impact of the Sound-Lab speaker is unquestionably that of a SOTA contender. Extremely transparent, very low in typical loudspeakerish colorations, limited only by the amplifier in dynamic headroom, and fast enough on top not to make us yearn for the Pyramid ribbon tweeter, the R-1 is on balance our preferred reference speaker as we go to press. An occasional touch of hardness or edginess intrudes on certain kinds of program material, but with the right amplifier and the right placement of the panels it isn't very obvious, certainly not enough to make us prefer the worse flaws of other expensive speakers.

One little fly in the ointment was the reversed wiring of the input terminals on our samples. Red was negative and black positive, messing up the phase alignment of our biamped system until we caught the error. Without a pulse generator, a microphone and an oscilloscope it would have been very difficult if not impossible to catch; let all owners of early samples beware. The error has meanwhile been corrected, we're told. It just goes to show that nobody is perfect, not even the remarkable Dr. West and his obviously sophisticated little company.

Swallow CM70

Swallow Acoustics, England; imported by Sunrise & Company, 85 Columbia Street, Suite 19A, New York, NY 10002. Model CM70 compact monitor loudspeaker, $495 the pair. Tested samples on loan from importer.

This smallish bookshelf system has already attracted some surprisingly favorable attention in Great Britain and is just now coming into the U.S.A. It consists of a Dalesford 8-inch woofer and the KEF T27 dome tweeter in a sealed box (yes, it's another of those), with the apparent crossover at approximately 4 kHz. The two drivers are in phase for a change, but again it's our impression that the little T27 is being driven too hard. (See also our comments on the QLN I above.) Tone bursts show an unmistakable ring pattern at 3.9 kHz, right in the crossover region, and the net result is a very edgy and fatiguing sound, even if the first-minute impression is one of clarity and focus.

The bass is peaked up *a l'anglaise* at 75 Hz, with a Q of approximately 1.3 (going up to 1.5 with increasing drive levels); frequency response is otherwise reasonably flat all the way to 25 kHz; pulse shape retention is good down to a width of 0.3 msec only, but there's relatively little trailing garbage after pulse cutoff. None of this is particularly relevant in view of the speaker's nasty cutting edge, which must be eliminated before the CM70 can be considered an audiophile product in our book.

Vandersteen Model IIA

Vandersteen Audio, 1018 South Mooney Boulevard, Visalia, CA 93277. Model IIA floor-standing 3-way speaker system, $920 the pair ($940 east of Denver). Matching 6'' high metal stands, $60 the pair. Tested #3236A and #3237A, on loan from manufacturer.

This is a modification of the Model II described, and favorably reviewed, in our last issue. Since the basic design hasn't changed, let's just restrict ourselves to whatever is different in the 'A' version. For one thing, the dome tweeter is now an Audax, not a Peerless. The 8-inch woofer is also new and also by Audax; furthermore the tuning of the vented (passive radiator) enclosure has been changed and is now much closer to optimum, with its -3 dB corner at 40 Hz. The crossover frequencies have been slightly shifted. Best of all, the three drivers are now in phase, pushing and pulling together on all transients. We wish to take just a tiny bit of credit for nudging Richard Vandersteen in these directions.

The net result of these mods is that the Vandersteen has become the smoothest, best-focused, least colored, most balanced-sounding dynamic speaker system known to us, exceeded in transparency, definition and neutrality only by the top electrostatics. All this despite the fact that the delay between tweeter and woofer is too great to permit any kind of pulse shape replication. Obviously that's only one criterion of loudspeaker accuracy and possibly not the most important.

It must be emphatically added that the above ranking applies only in the case of music played at less than very loud levels. The speaker is basically a small-signal design, with clarity rather than dynamic headroom as its top priority. When pushed, not even terribly hard, it begins to run into trouble. The stress becomes quite audible; it isn't just a subtle loss of quality. On the lowest organ pedal passages, the woofer flirts with total breakup unless a firm hand is applied to the volume control. The midrange and the tweeter also have their very definite dynamic limits, determined in part by the choice of 6-dB-per-octave crossover slopes. The DCM Time Window is unquestionably better in this respect, though not as free of sonic colorations.

All things considered, even the last-minute increase in price as we go to press, the Vandersteen Model IIA is our inevitable choice for the speaker end of our ''Reference B'' system, since it represents a fair approximation of what we consider accurate sound reproduction, in contrast to comparably priced speaker systems with a sonic personality of their own.

Recommendations

Interesting new speaker systems are waiting in the wings (such as for example the Dayton Wright XG-10, the revised Beveridge System 3 and the Pyramid Metronome Model 3), so don't carve any of this in marble yet. But here's how we see it currently.

Best speaker system: Sound-Lab R-1, biamped with Janis W-1 subwoofer (see also Reference A update in this issue).

Best speaker system at a three-figure price: Vandersteen Model IIA (see also Reference B update).

Best speaker value per dollar: DCM Time Window.

* * *

Best tweeter: Pyramid Model T-1.

Best subwoofer: Janis Model W-1 with Interphase 1.

Best subwoofer per dollar: The Bass Mint Model 10/24.

A Genuine Breakthrough in Inexpensive Integrated Amplifiers

By the Staff of
The Audio Critic

How about a $175 integrated amplifier that wipes out most $1000 preamps plugged into most $1000 power amps, except maybe in sheer wattage? Here's conclusive proof that good, realistic thinking in audio design has nothing to do with dollars.

There's no reason on earth why an integrated amplifier shouldn't be just as good as any preamp/power-amp combination. Electrons have no perception of whether the devices they're flowing through are all on one chassis or on two chassis. All compromises associated with integrated amplifiers in the audio purist's mind have to do with the history of hi-fi marketing rather than technical limitations. Historically, cheap and dirty audio amplification has been packaged this way, but that doesn't mean there's anything inherently wrong with the format itself.

As a matter of fact, the integrated format is potentially your best assurance of a correct interface between the preamplifier and power amplifier sections, assuming that the designer knows what he is doing. Consumers who don't know what they're doing may end up, for example, with a superfast (high slew rate) preamp feeding a much slower power amp that has little or no filtering at its input. The result then is a classic case of TIM. It's also easier to pick up RFI when there's a long cable between the preamp output on one chassis and the power amp input on another, separate chassis.

When it comes to doing a really good job for as little money as possible, the integrated amplifier is of course the only way to go. One power supply instead of two, a lot less metalwork, fewer circuit boards, less wiring—it all adds up to a considerable saving without necessitating any sacrifice in performance as compared to an equivalent two-chassis design. Only the audiophile image suffers, but image is a product of the marketplace, not logic.

Needless to add, all comments on preamplifier and power amplifier testing in this issue are equally applicable to integrated amplifiers. No further introduction is therefore necessary.

NAD 3020

NAD (USA), Inc., New Acoustic Dimension, Mackintosh Lane, PO Box 529, Lincoln, MA 01773. Model 3020 Integrated Stereo Amplifier, $175. Two-year warranty. Tested #3225220, on loan from manufacturer.

It looks unassuming rather than cheap—a simple black box with a full complement of controls, including bass and treble, as well as a five-LED peak-power indicator monitoring both channels and displaying the higher output at any instant. The LED's are labeled 1, 5, 10, 20 and 35 watts into 8 ohms; the last is about 2½ dB above the obviously ultra-conservative 20/20-watt continuous power rating.

The unit came to us very highly recommended, so we threw the toughest test at it right up front. With a variety of speaker systems, we A-B-ed it against our very best preamp/power-amp combination, the Cotter System 2 feeding the Rappaport AMP-1. (The latter has meanwhile become extinct.) The price ratio of A and B in this test was roughly 15 to 1. Well, what can we tell you? Everyone who was listening agreed that the NAD wasn't as good. Everyone also agreed that the difference was amazingly small. Both signal paths sounded clean, transparent, unstrained and musical. The NAD 3020 had a somewhat less open, neutral and finely detailed sound; it clipped a bit sooner; nevertheless, it wasn't really a letdown to switch to it because it was completely free of the hard, "electronic" quality of most transistor amplifiers, cheap or expensive. If the Cotter/Rappaport combination hadn't been available then and there as a reference, the NAD would have been accepted as just right—that's how good it is. By itself, it's difficult to fault it in clarity, smoothness and just plain accuracy.

We were able to make further and more detailed listening comparisons, since the 3020 can be separated into its preamp and power amp sections via jacks in the rear. Thus it can be inserted into a reference system either as a preamp or as a power amp and A-B-ed against others. What we found out about it that way is equally impressive. The preamp section ranks just below the top five or six separate preamplifiers we've tested so far (at any price!) and doesn't sound dramatically inferior to any of them. It never gets hard or overbright and is just a tad short of the ultimate in transparency. If the RIAA equalization were more accurate, we could almost begin to talk about "Reference B" quality. As it is, the error curve drops to −1 dB at 20 Hz, bumps up to +0.2 dB at 430 Hz, and shows a gradual decline above 1 kHz, down to −0.7 dB at 20 kHz in one channel, −0.4 dB in the other. Not too bad, but not excellent. The power amplifier by itself is perhaps even more remarkable; next to the Hafler DH-200, for example, it sounds a little compressed and less open but also smoother and sweeter, without any trace of that hard glint on top. In other words, it isn't totally surpassed by the Hafler, which in turn is surpassed by only six or seven other power amps known to us, at any price. For a $175 amplifier with a free preamp thrown in, that's not bad at all.

The subjectively perceived dynamic headroom of the 3020 can be increased by switching in the "soft clipping" feature, of which NAD appears to be inordinately proud. In our opinion, this is a double-edged gimmick that takes some of the unpleasantness out of frequent clipping when the amplifier is being pushed but also impairs the depth and three-dimensional detail of the reproduced sound. Our high rating of the NAD 3020 is based on its sonic quality with the soft clipping switch in the off position.

The most interesting question, of course, is how NAD is able to do so much for so little. What do they know that others don't? New Acoustic Dimension is an international organization, originally founded and financed by a group of dealers, with offices in several countries and production facilities in Taiwan. Being dealer-based gives them a realistic outlook on consumer needs; having access to reasonably skilled labor at relatively low cost gives them an edge in price. The 3020 isn't built like a Mark Levinson amplifier but it uses parts of fairly decent quality in all the important places and makes a few compromises wherever the penalty is tolerable. The designer of the entire line is Bjorn-Erik Edvardsen, a Norwegian now living in London, who has some very strong convictions about spending the available production budget on sound rather than cosmetics and sales features. He also seems to have a set of highly intelligent and effectual priorities in circuit design, giving us further evidence in support of our long-standing conviction that good thinking costs no more than bad thinking.

We were fascinated to find, for example, that the 3020 is not only bandwidth-limited to reject infrasonic and ultrasonic garbage but also happens to use high-pass and low-pass characteristics that are very similar to those of the state-of-the-art Cotter NFB-2 filter/buffer. Not that the Cotter filter's highly sophisticated time-domain correction is entirely duplicated, but the magnitude of the low-frequency roll-off is about the same, and the measured rise time of 9 microseconds is *exactly* the same. What a coincidence and what a corroboration! DC-to-light freaks, eat your hearts out. Correctly bandwidth-limited systems simply sound better. Large output transistors that are just coasting most of the time, not much feedback, a very carefully designed power supply, and no current-limiting protective circuitry are some of the other plausible reasons of the 3020's sonic success. Without any allowance for its low price, this must be considered a thoroughly modern amplifier, designed with total awareness of the errors of the past and obviously capable of handling complex speaker loads with aplomb. We're impressed beyond our wildest expectations.

The one thing that remains to be seen is whether or not the NAD 3020 will perform as impressively after years of heavy use as it does when it's new. We gave our sample as much of a beating as we could and found no change taking place, but we can't make any unqualified promises. It just isn't a mil-spec amplifier. It would be a pity, though, if all the $1000 preamps and $1000 power amps that are better built but don't sound nearly as good outlived it to pollute the ears of our children.

NAD 3045

NAD (USA), Inc., New Acoustic Dimension, Mackintosh Lane, PO Box 529, Lincoln, MA 01773. Model 3045 Integrated Stereo Amplifier, $350. Two-year warranty. Tested #3459341, on loan from manufacturer.

This one is rated at 45/45 watts into 8 ohms, costs twice as much as the 3020 above, and has a number of extra features, including VU-style power meters on the front panel. It represents, however, an earlier phase of NAD's circuit design philosophy and, in our opinion, doesn't quite achieve the same sonic transparency and smoothness. Next to the 3020, it sounds a tiny bit harder, brighter and less clean—but just. Furthermore, unlike the 3020, it can't be separated into independent preamp and power amp sections. It does have 3½ dB more power, though, when you need it and is otherwise very much in the same mold as the 3020.

We understand that the 3045 is due for early replacement by a new NAD amplifier of comparable power rating that will incorporate everything that's special in the 3020 and then some. Until then this remains a truly excellent product whose only fault is that the 3020 sounds even better and costs less. Had the 3045 been the only NAD amplifier sent to us for testing, our review of it would have been almost as enthusiastic as the one above, with only very minor reservations. Too bad, but it's the old story—the best is the enemy of the good.

The New Generation of Power Amplifiers

By the Staff of
The Audio Critic

It doesn't look as if anybody really knew what makes a power amplifier perfect, but through sheer tenacity a few designers are getting within striking distance.

Now hear this. Forget about THD measurements. Forget about SMPTE IM. Forget about CCIF IM. Forget about slew rate. Forget about bandwidth. Forget about square wave response, into any kind of load, unless it's truly horrendous. Forget about "black box" tests (comparing the output against the input) in general. They just won't tell you what you want to know about power amplifiers. Sure, they'll identify specific malfunctions and crude design errors; they'll separate the dogs from the half decent ones. But if you're after the best sound that money can buy—or the next best sound that a lot less money can buy—forget it.

As you know, we've had our reservations about black box tests for several issues now; with this latest batch of power amps we've reached a firm conclusion. There just isn't any sonic correlation. We still put all power amps through our routine bench tests, mainly to get some idea of the design philosophy behind them and to eliminate the possibility of attributing audible effects to nonexistent technical causes, in the manner of certain underground reviewers. But we certainly don't claim at this point that we can predict the sound of a power amplifier from its measurements, even approximately. If nothing else, the Bedini Model 25/25 (see review below) has cured us of that illusion. All our evaluations are therefore based on how the amplifier sounds when inserted into our "Reference A" and "Reference B" systems.

Feedback falls into further disrepute.

Shortly before press time we received a devastating document that appears to confirm our worst doubts about standard methods of amplifier measurement, not to mention amplifier design in general. It's the preprint of a technical paper to be delivered by Dr. Matti Otala on February 25, 1980, at the 65th Convention of the Audio Engineering Society in London.

The paper presents rigorous mathematical proof, for the most generalized, all-inclusive case, that feedback cannot make amplifier distortions go away; all it can do is to change one kind of distortion into another. By the application of feedback, the amplitude nonlinearities of the open loop are converted into phase nonlinearities of the closed loop. That's all. The garbage cannot, by definition, be made to disappear; it's simply swept into another corner. In the typical feedback amplifier, the amplitude of the audio signal phase-modulates the high-frequency components of the signal. Furthermore, any amplitude intermodulation distortion in the open loop is converted into phase intermodulation distortion in the closed loop. What about TIM, alias SID? It turns out that it (he?) is a limit case of this feedback-generated phase modulation effect, with all shades of gray possible before the actual black eruption occurs. None of this shows up on standard tests. Scary, isn't it? Dr. Otala promises further papers on the audibility of the effect and on a method to measure it, adding that it seems to be particularly annoying in amplifiers that use high values of feedback to suppress crossover (notch) distortion.

This is no small matter; once the analysis and the conclusions are familiar to the audio engineering community, amplifier design will never be the same again. We wonder how prompt the Audio Engineering Society will be to publish the paper in the *Journal*, since there are many AES members with a vested interest in high feedback and triple-oh amplitude distortion figures. We trust that scientific objectivity will prevail. Meanwhile we feel justified in many of our previously expressed but not nearly as rigorously derived opinions and relieved that the best minds in the business are also having trouble finding measurement methods that correlate with listening.

Amber Series 70

Amber Electronics, Inc., 917-B Preston Avenue, Charlottesville, VA 22901. Series 70 Power Amplifier, $459.95. Three-year warranty. Tested #700100 (also two earlier, not yet optimized samples), on loan from manufacturer.

There isn't much we can say about this solidly built, sensibly designed, conservatively operated 70/70-watt power amplifier, except that we discern in it no obvious errors of concept or execution and that it sounds better to our ears than any other amplifier in its class, including the Hafler DH-200 and the PS Model One, though not by a wide margin.

You may think that's a pretty heavy statement, but it isn't really, not in a product category where things are changing very fast these days and mostly for the better, and especially not since the Bedini Model 25/25, for only $190 more, offers considerably superior sound, though not as much power. Furthermore, our ranking of the Amber applies only to the very latest production version, incorporating certain circuit modifications without which the amplifier doesn't sound nearly as good, and then only when its speaker fuses are changed to 5 amps or heavier. (See also the preamble to the speaker reviews in this issue.) Thus modified and fused, the Amber sound very smooth, open, neutral and clearly detailed, with a good, solid bottom. By comparison, even the best sample of the Hafler is a little zingier on top and the PS perhaps a wee bit coarser in grain—but just. On balance, the Amber is our new power amplifier choice for "Reference B."

One feature that makes the Amber even more desirable and certainly more versatile is the built-in capability to be strapped for mono operation. At the flick of a switch, you have a 200-watt (into 8 ohms) mono amplifier that sounds just as good as the stereo channels, even though the strapping is accomplished in a theoretically less pure way than the classic balanced bridge connection, which requires accurate phase inversion.

How do we account for the good showing of this first product of a new and little-known company? It seems that they have their act pretty well together. Even in physical appearance, the Amber looks much more finished and professional than what we expected. Really nice. The power supply is also considerably more generous than what is generally seen in this price and power category; the audio signal path appears to be simple, straightforward and trouble-free. Perhaps basic good sense, no penny-pinching, and the avoidance of vulgar design errors will get you further in product development than half-baked originality.

Audionics BA-150

Audionics, Inc., Suite 160, 10950 SW 5th Avenue, Beaverton, OR 97005. BA-150 Analog-Digital Stereo Power Amplifier, $2950. Three-year warranty. Tested #10163, on loan from manufacturer.

This one is a whole bucketful of paradoxes. First of all, in immediate contradiction of our generalizations above on amplifier measurements, black-box testing does offer some clue to the sound quality of the BA-150. Secondly, its enormous price tag, although promising more than what the amplifier delivers in audible benefits, does have some justification in the beautiful and thoroughly professional construction. Thirdly, the design looks nostalgically backwards, to the good old days of vacuum tubes and output transformers, and at the same time aggressively forward, to bias and balance regulation with CMOS digital logic. Finally, with three different choices of feedback and three output impedance connections, the BA-150 has nine different kinds of sound—but we're getting ahead of ourselves.

Designed by David Berning of the National Bureau of Standards (as a private project, of course, not for the NBS), the BA-150 is a hybrid design, with a pair of 6LF6 tubes in a new Class B configuration driving each speaker through an output transformer, and with both tubes and solid-state devices used in other parts of the circuit. The aforementioned CMOS digital logic circuitry continuously monitors the operating characteristics of the output tubes and automatically adjusts biasing for optimum performance. Either 150 watts per channel or 40 watts per channel (for speaker protection) can be chosen by means of a front-panel switch; seven different colored lights provide information at a glance as to what the various parts of the amplifier are doing. It's a neat and reassuring package.

The output transformers are tapped for 4, 8 and 16-ohm loads; in our opinion, a combination of series and parallel strappings, utilizing all windings of the secondary under all loading conditions, would have been a more elegant and better optimized solution. (Remember the old Partridge output transformer? You're no spring chicken if you do.) Another aspect of the design that we aren't convinced about is the feedback selector switch, which allows the user to decide whether to apply 0, 8 or 14 dB of feedback around the open loop. After three minutes of listening, it becomes quite apparent that 8 dB is the right figure, with just about any speaker load, neither 0 dB nor 14 dB giving anything even approaching the same clarity and freedom from colorations and/or stress. Obviously, there are audible amplitude distortions with no feedback and audible time-domain problems with too much feedback, leaving 8 dB as the right way to split the difference. It should have been the built-in design choice, without giving anyone the opportunity to mess it up. With the three positions of the feedback control and the three output taps, there are nine possible combinations, each of which sounds totally different from the other eight. The cleanest, most transparent-sounding combination is probably 16 ohms out with 8 dB of feedback—but what if you're driving speakers that drop to 3 or even 2 ohms somewhere in their range, as so many do?

At its best, the sound of the BA-150 is amazingly sweet, smooth and edgeless, often in dramatic contrast to that of the transistor amplifier used in a particular A-B comparison. The initial listener reaction is always very positive; only after a few minutes of exposure does it become apparent that there's always a certain amount of sonic mud, thickness and veiling of detail. Just this once, we believe we can offer some laboratory data to corroborate our ears. For one thing, the amplifier produces a tremendous amount of IM distortion. With 0 dB feedback, the SMPTE figures are ridiculous: 5.4% at 30 watts into 8 ohms; over 10% at 100 watts into 8 ohms. With 8 dB of feedback, these figures drop to 2.35% and 4.2%, respectively. Into 16 ohms, we measured a halfway respectable 1% at 30 watts (again with 8 dB of feedback);

the question is, where does the relative forgivingness of the human ear to amplitude distortions reach its limit? These numbers are too large for comfort. Furthermore, the amplifier is very, very slow. Rise time with an 8-ohm resistive load is between 5.5 and 7 microseconds, depending on the setting of the feedback selector; with capacitive loads, considerably worse low-pass effects can be observed, greatly varying with the combinations of transformer taps and feedback settings. In a worst-case situation, still using real-world capactive loads as encountered in electrostatic speakers, square waves change into virtual sine waves. This is the first power amplifier with which we hestitate to recommend the use of the Cotter NFB-2 noise filter/buffer, since the rms sum of the rise times of all the signal-path components in series might then exceed the threshold of hearing, so that the slowing down of the signal would be definitely audible. Unfortunately, this filtering effect of the BA-150 takes place at the output, not the input, and thus the benefits of the Cotter filter aren't necessarily duplicated. What all this boils down to is that lots of IM distortion combined with borderline speed on transients may very well be responsible for the sonic personality of the amplifier.

Even so, we're convinced that there exist installations in which the power, rugged construction, virtually certain reliability and velvety sound of the BA-150 will make it the right choice for a particular user, high price notwithstanding. We continue to prefer, however, a few solid-state power amplifiers that offer comparable smoothness in combination with greater clarity and better resolution of inner detail.

Audire 'Crescendo'

Audire, Inc., 18474-E Amistad Street, Fountain Valley, CA 92708. 'Crescendo' Power Amplifier, $350. Three-year warranty. Tested #5070 and #5129, on loan from manufacturer.

For $350, this surprisingly well-built 60/60-watt power amp gives you a midrange that's close to SOTA in openness and transparency. In fact, when we first heard it, we were reminded of the midrange quality of the Rappaport AMP-1. At the same time, the 'Crescendo' sounds quite hard and edgy on top, though a little less so in the second sample we auditioned, which had been a bit more carefully adjusted for correct bias as a result of our complaints about the first. Even this less irritating sample, however, was clearly not as smooth and neutral in the upper frequencies as the Hafler DH-200, which itself is a small step or two from the ultimate in this respect.

On the laboratory bench we observed some square wave anomalies even with a purely resistive load, not to mention quite a bit of overshoot and ringing when capacitance was added, but we no longer set much store by these tests. Most probably the cutting edge we hear has to do with the operating characteristics chosen for the various devices in the circuit, hence the variability with small bias adjustments.

One nice feature at this price is the very graphic power output indicator; it flashes twelve LED's per channel, in green, yellow and red. We could watch it for hours, but we'd rather listen to the Amber, PS or Hafler, all of which admittedly cost a little more. We call this one a near miss.

Audire DM 700

Audire, Inc., 18474-E Amistad Street, Fountain Valley, CA 92708. DM700 Power Amplifier, $1400. Three-year warranty. Tested #7078, on loan from manufacturer.

We expected a lot from this one, the flagship of the Audire line, rated at 350 watts per channel into 8 ohms (that's right, three-five-oh, or 500 watts into 4 ohms) and presumably designed with the same understanding of what a transparent midrange is all about as the 'Cresendo' reviewed above. As we learned, it does share the latter's midrange qualities (excellent!) but also its hard, overbright character further up in the audio range. We found our sample of the DM700 very fatiguing to listen to for any extended length of time; the zinginess was much too aggressive.

After a while we began to suspect incorrect bias settings, as in our first sample of the Crescendo, so we made an attempt to readjust the bias. What we didn't know was that the bias pots in the DM700 don't just operate within the allowable bias range but can actually be rotated all the way to zero resistance. (This saves the price of a protective resistor in series with each pot.) As a result we zapped one of the boards with too much current, and it went up in smoke. End of our tests, as El Cheapo strikes again.

Square wave tests, just before this fiasco, revealed some of the same strangenesses we had observed in the Crescendo. No big deal. It should be noted that the huge output of the DM700 is obtained by means of a bridge circuit, which in some ways is also a money-saving gimmick (four times the power into the same load by just doubling the number of output devices, at the expense of various other important design considerations); nevertheless, we could certainly use and would welcome this kind of power any way we can get it, as long as the sound remains sweet and smooth. One thing the DM700 does right is the use of entirely separate power supplies for each channel; it's an amplifier with considerable appeal and potential at this price, one of those we would like to like better. A modified sample has been promised to us and is supposed to be on its way as we go to press; look for a follow-up review in the next issue.

Bedini Model 25/25

Bedini Electronics, 13000 San Fernando Road, #9, Sylmar, CA 91342. Model 25/25 Class A Power Amplifier, $650. Tested #250029, on loan from manufacturer.

Here's the sleeper of the year: a 25/25-watt (14 volts out) little Class A unit that performs most unimpressively on the lab bench but sonically wipes out all other amplifiers known to us—in transparency, depth of perspective, definition of inner detail *and* smoothness of highs! Its only rivals are the now defunct Rappaport AMP-1 and a last-minute arrival, the JVC M-7050, both of which are in a totally different power and price category. We said rivals, mind you, not equals. The whole thing is rather hard to believe.

John Bedini claims that his circuit is "pure Class A" rather than "sliding Class A" (a la Threshold, etc.); we aren't sure whether his definition of "pure" would be accepted by the designer of the Mark Levinson ML-2 Class A amplifier, for example, but that's beside the point. Technical hairsplitting isn't the long suit of the Model 25/25. In fact, hairsplitting laboratory tests will reveal that it suffers from rather severe slew rate limiting; we strongly recommend the use of the Cotter NFB-2 noise filter/buffer with this amplifier to prevent it from getting zapped with superfast rise times devoid of auditory information. It will function a lot more happily that way. But that's not all. We could hardly believe our eyes when we passed square waves through the Model 25/25 into an 8-ohm resistor, with an additional 2-microfarad capacitor connected directly across the output terminals. The amplifier went into such severe oscillation that it almost immediately blew both 8-amp fuses in its power supply. *Mamma mia!* Since this test load is a fairly accurate model of what is presented to the amplifier by certain electrostatic speakers, the relatively small series inductance of a length of speaker wire acquires crucial importance as protective isolation in an actual music system utilizing the Bedini. It has no equivalent built-in protection. (See also the speaker wire article in this issue. Incidentally, nothing makes the highly capacitive Quad electrostatic sound quite as perfect as the Model 25/25—with a bit of wire in between, of course.)

How come this test-bench turkey sounds so beautiful in the real world? Ah, if we knew that, we could devise real-world oriented bench tests in which the Bedini would obviously not come out a turkey. Our best guess is that all the solid-state devices used in the amplifier are deliriously happy at their particular operating voltages, currents and temperatures; those big output transistors are just coasting; nothing is stressed beyond the most comfortable linearity; and whatever little glitches exist are amplitude modulations rather than time modulations. At least with the Cotter filter at the input and a nice length of mildly inductive speaker wire at the output, right? In addition, the amplifier can put out a fair amount of current, even though it's limited in voltage output, so it sounds more powerful driving complex speaker loads than a typical 25-watter. And that's the best we can do on that one, folks.

Anyway, while trying to figure out what makes the Bedini Model 25/25 tick, we're shamelessly enjoying it in our "Reference A" system, replaced by the JVC M-7050 only when we want to play something very, very loud.

Bedini Model 45/45

Bedini Electronics, 13000 San Fernando Road, #9, Sylmar, CA 91342. Model 45/45 Class A Power Amplifier, $1300. Tested #450047 (AC coupled) and #450053 (DC coupled), on loan from manufacturer.

This is supposed to be a scaled-up version of the Model 25/25, with everything essentially the same except the bigger power supply. Well, there's one other thing that isn't the same in our opinion, and that's the sound. The Model 45/45 isn't even unequivocally superior to a good sample of the Hafler DH-200, at one third the price, let alone the smaller Bedini or the JVC. Where the Model 25/25 is utterly smooth and edgeless, the 45/45 exhibits that characteristic little transistory zing and hardening, and its midrange transparency and delineation of high-frequency detail are merely good, not great. Obviously, there's something going on in the circuit that even John Bedini hasn't quite got a handle on at this point. We do hope he gets there before his coming 200/200-watt model is finalized; with that kind of power and the sound of the *little* amplifier, we could all just wash up and go home. A 100/100-watt version has been heard by a number of people whose ears we trust, and they report that it isn't quite in the same class sonically with the 25/25.

Very early samples of the Model 45/45 had huge output capacitors (a la Futterman); we see nothing wrong with such AC coupling, but the current production model has been changed to the DC-coupled configuration of the entire line. We could hear no significant difference between the two versions. The test-bench behavior of the 45/45 is very similar to that of the 25/25, and you already know how relevant that turned out to be. Wouldn't it be nice if we all knew as much about measuring amplifiers as we did only a few years ago?

Hafler DH-200
(follow-up)

The David Hafler Company, 5910 Crescent Boulevard, Pennsauken, NJ 08109. Model DH-200 Stereo Power Amplifier, $429.95 wired. (In kit form, $329.95.) One-year warranty. Tested successive production samples and modifications, all on loan from manufacturer.

Our enthusiastic recommendation of this most attractively priced 100/100-watt MOS FET power amplifier in the last issue has created something of a credibility problem for us. Early production units turned out to be not nearly as excellent as the preproduction sample we had tested in good faith, having been solemnly assured by the manufacturer that the final production version would be electrically and sonically indistinguishable. Let's straighten out the record by tracking through the entire sequence of events following our review.

The first thing that happened was that the designer of DH-200, Erno Borbely, left The David Hafler Company to become head of the European training program of National Semiconductor Corp. Ed Gately, Hafler's president and an excellent engineer in his own right, was left with the problem of following through on the DH-200 project, without being on intimate terms with every minor component in the circuit. Somehow or other, perhaps as an untested afterthought by the departing designer or on someone else's instructions, a high-beta transistor was substituted in some production runs for the low-beta device used in certain low-level stages of the circuit. These DH-200's, all of them having serial numbers below 3939000, sounded hard, zingy and dimensionless. The low-beta transistor was then reinstated in all applicable positions. Later it was discovered that certain transistors had to be more carefully matched for best results; still later it was decided that certain capacitors had to be replaced in order to eliminate some remaining vestiges of zinginess. By the time we tested production samples in the 3946000 series, some were already better than our original review sample, though not all. Finally we got a modified sample that looked forward a couple of months; it was supposed to incorporate all the minor capacitor changes and other little tweaks that would be in production early in 1980. This one was definitely cleaner and sweeter than any previous version we had tested, including the preproduction one. Meanwhile, however, we still kept hearing about the variable quality of the DH-200's being received by dealers.

If the Hafler were still out top choice in the "Reference B" category, we would probably explore the situation still further. Our latest listening comparisons show, however, that both the Amber Series 70 and the PS Model One are marginally superior even to this "final" version of the DH-200. The latter still has a tiny bit of that treble shimmer or zing we pointed out in our original review; the Amber and the PS are both a shade better in this respect and also have a more solid bottom end, as well as a slightly more transparent midrange. Since they all cost about the same (the DH-200 has gone up in price by $30), there's not much point in agonizing over the possible variability that may or may not exist in the latest production runs of the Hafler. If you happen to own a DH-200 that isn't quite up to snuff, you probably aren't just imagining things and should holler until your dealer or Hafler fixes it free of charge.

None of this means that the DH-200 isn't still an outstandingly fine audio component; in fact, we're convinced that, other things being equal, a MOS FET power amplifier is inherently superior to one that uses bipolar output transistors. But other things are seldom equal.

JVC M-7050

US JVC Corp., 58-75 Queens Midtown Expressway, Maspeth, NY 11378. M-7050 Stereo Power Amplifier, $1500. Two-year warranty. Tested #13400021, on loan from manufacturer.

This one came in so close to press time that we didn't even have time to put it on the lab bench. All we could do was to have it in our reference system for a week or so, listen to it as much as we could, write this brief last-minute review, and insert a word or two in the other reviews where a comparison with the M-7050 seemed essential. A follow-up in the next issue should take care of the loose ends, including measurements (if at all relevant.)

Even such brief exposure reveals outstanding qualities in the M-7050. Here's one Japanese amplifier that doesn't sound Japanese. Its basic sonic character is in the mold of the most sophisticated American high-end electronics. At this point, we rank it immediately below the Bedini Model 25/25 in transparency, lack of "personality," smoothness, and delineation of detail. Just about all other amplifiers sound coarse, veiled or edgy next to the JVC, at least a little bit. And here comes the punch line: the M-7050 is rated at 150 watts per channel into 8 ohms! It's the only "monster amplifier" we've tested so far that can hack it in the purist category.

The output stage of the M-7050 is biased for what JVC calls Super-A operation; it isn't pure Class A but it appears to be more linear, judging by the sound, than Class AB amplifiers we're familiar with, and the power efficiency is still reasonable. The amplifier weighs about 50 pounds. Two neat features are the huge power level meters and the switching facilities for A-B-ing two pairs of speakers (or connecting both pairs simultaneously). For further discussion and a more finely tuned evaluation, see the next issue.

PS Model One

PS Audio, 3130 Skyway Drive, #301, Santa Maria, CA 93454. Model One Power Amplifier, $449. Five-year warranty. Tested #0397, on loan from manufacturer.

Another very late arrival that we just had to mention before going to press, the 80/80-watt PS also proved itself amazingly quickly against all amplifiers in its power and price category—and then some. After somewhat limited exposure to it, we rate it just a hair below the Amber Series 70 and an equally small distance ahead of our best sample of the Hafler DH-200.

The PS III phono preamp already demonstrated to us that these people know what totally neutral, uncolored audio amplification is all about, and their first power amp now clinches the proof. It has no personality. The bass is deep and solid, the midrange is impressively transparent, the highs are clean and nonirritating, spatial perspective is excellent. The Amber has perhaps a slightly more refined and suavely detailed sound; the Hafler is a wee bit zippier on top and maybe not quite as firmly controlled and focused; essentially all three are in the same class, and a very classy class it is.

PS claims that a special, patent-pending circuit they call Storage Linearized Amplifier is responsible for the good sound. The amplifier came in much too late for us to have an opinion on that, but any awareness of storage as a cause of audible time-domain perturbations earns an automatic "right on!" from us. This is clearly an audio product that has been engineered *and* listened to.

Rappaport AMP-1
(epitaph)

In case you haven't heard, Andy Rappaport is out of business. A. S. Rappaport Co., Inc., has declared bankruptcy; Andy has a salaried job in Boston with a high-technology company that has nothing to do with audio; his plan is to remain an audio designer and consultant in his spare time but never again a manufacturer. All a case of too much too soon with too little capital, we'd say.

This creates a problem for those who own Rappaport equipment, some of which is rather temperamental and requires expert servicing. The AMP-1, for example, has an almost 100% failure rate, if you count minor but annoying malfunctions (hum, excessive DC offset, asymmetrical clipping, overheating, etc.) as failures—which they are. Andy informs us just as we go to press that an "official" servicing agency is about to be set up in a New York suburb, which will have his full cooperation and the benefit of his engineering advice. Repairs, however, will be charged for; the warranty died with the bankruptcy. Quite independently, the largest Rappaport dealer in the New York area has declared to us his intention to honor the warranty and provide free service to his own Rappaport customers (but not to any other dealer's), probably by going through this same agency. So all is not lost, even if you didn't heed our warning about reliability and gambled on the Rappaport sound.

That sound, incidentally, hasn't been clearly bested yet; the Bedini Model 25/25 is the only power amplifier we know of that may quite possibly be superior to the AMP-1, but we'll never know for sure, having bailed out of the two we owned when the news of the dissolution reached us. The JVC M-7050 is probably comparable but not better.

All we can say at this point is that we never imagined it would end this way. Even so, it isn't our intention in the future to run a Dun & Bradstreet credit check on every bright young man who sends us an amplifier for review. We've got our hands full with just the performance aspects of consumer protection.

Sonotron PA-2000

Sonotron A/S, PO Box 2114, N-7001 Trondheim, Norway. PA-2000 stereo power amplifier, $650 (approx. export price, not retail). Tested #1189, on loan from manufacturer.

We don't quite know what to make of this 100/100-watt Class AB amplifier from Norway. It's a rather attractive package; the parts appear to be of good quality; even the price seems to be within the bounds of reason for a European high-end product. The reputation of the amplifier is quite high in Northern Europe. And yet—we just can't get decent sound out of the PA-2000.

On every little dynamic peak at the higher frequencies—strings digging in, sopranos hitting a high note, trumpets punctuating a phrase—the amplifier literally protests, creating a hot, piercing spike of sound that goes well beyond the typical hard or edgy quality we often complain about. It's extremely fatiguing and makes the amplifier virtually unlistenable.

In this case we really expected to see some kind of ringing on square waves or other obvious transient anomalies when we put the amplifier on the test bench. We were wrong. The Sonotron passes all routine black-box tests with a clean bill of health. In fact, it looks very, very good on all conventional spec-sheet type measurements. We now suspect that what we heard has to do with feedback-related time modulations, which are very hard to catch on the wing and display on meters and CRT's.

While we were in the middle of all this, a Scandinavian audio dealer happened to mention to us that the Sonotron sounds much better when thoroughly warmed up. We routinely warm up all amplifiers for a good many hours before we listen to them critically, but this time we took special pains, even attempting to overheat the amplifier by preventing the circulation of air around the heat sinks. It made no audible difference, however, and we reluctantly gave up.

Recommendations

We still haven't been able to get our hands on a Mark Levinson ML-3 (Class AB, 200/200 watts), so we'll just have to take Mark Levinson's word that the less powerful ML-2, which we tested in mid-1978, is a more accurate amplifier and hasn't changed all that much since. We haven't tested the very latest from Audio Research or Threshold, either; otherwise we have the field reasonably well covered, we feel. We'll keep trying to test all the most promising contenders.

Best-sounding power amplifier tested so far, regardless of price, but of limited output capability: Bedini Model 25/25.

Best-sounding power amplifier of large output capability: JVC M-7050.

Best power amplifier per dollar: Amber Series 70 (see review above for qualifications).

Important Announcement from the Publisher

The last few issues of ***The Audio Critic****, including this one, have been in effect double issues for the price of one—and also double issues in the time it took to publish them. We can no longer go on like this.*

Charter subscribers will recall that ***The Audio Critic*** *was originally a publication of more limited contents that came out more often, at only a 6.7% lower subscription price per issue. Instead of adjusting for the most devastating inflation in recent history, we've been writing, producing and pricing issues as if costs were lower than ever and money easier to come by. This must come to a halt, right now.*

The next issue will contain fewer reviews and features than this one and won't take nearly as long to be published. Our mail indicates that most of our subscribers would rather pay the same price per issue for smaller issues more often. In the course of 1980, we plan to convert to an entirely new format, which will eventually consist of quite thin issues published with considerably greater frequency, plus one very big, fat and highly educational reference issue, a kind of semipermanent handbook to which the thinner issues will keep referring. Full details will be announced in the next issue; there may have to be another price increase, alas; however, the six-issue subscription cycle will continue, only the page allocations will be different. The total six-issue package, including the reference handbook, should be richer in contents than ever.

It must also be pointed out that the cover date of the present issue refers to the period during which the equipment tests took place; revisions, updates and last-minute additions went well into the first two months of 1980, however, so you can consider all the information presented here to be thoroughly up-to-date, even to the extent that some of the equipment reviewed is barely available yet as this issue is about to be mailed to you.

Further Developments in Step-Up Devices for Moving-Coil Cartridges

By the Staff of
The Audio Critic

The latest crop includes an almost-SOTA transformer at an attractive price, a major improvement in what was already the best-per-dollar pre-preamp, plus the usual quota of new mediocrities.

After our experience with the Audio Standards MX-10A (see pages 62 and 63 of the last issue), we have the deepest suspicion of any kind of laboratory test of a pre-preamp or transformer. That was as close to a straight wire with gain as we had ever seen on our lab bench, and it didn't mean a thing. We tracked through the same tests on the present batch of step-up devices, but whatever we observed has little bearing on our conclusions as reported in these reviews. Only in the case of the RWR transformer (see below) was there any correlation between a particular electronic test and an audible effect. Even in that instance we can't be absolutely sure.

All of the listening tests were performed by inserting the step-up devices one by one into our "Reference A" system (see updates in this issue). The slight differences in gain introduced this way were compensated for by careful calibration of the main volume control of the system. No A-B switching system was used, as we know of none that will work properly at MC cartridge output levels.

Denon AU-340

American Audioport, Inc., 1407 North Providence Road, Columbia, MO 65201. Denon AU-340 Audio Step-Up Transformer, $410. Tested sample on loan from importer.

This is an attractive and obviously well-made transformer, with switchable inputs for two separate pickups, front-panel choice of 3-ohm or 40-ohm connection, plus bypass. Sonically it rates fairly high but is no match for either the Cotter or the RWR. By comparison, it's a little edgy, especially on strings, and also has a relatively unobtrusive but definitely perceptible midrange coloration. If nothing better were available, the AU-340 would have to be considered quite clean and transparent, however.

Denon HA-1000

American Audioport, Inc., 1407 North Providence Road, Columbia, MO 65201. Denon HA-1000 MC Cartridge Head Amplifier, $440. Tested #110157, on loan from importer.

Another solidly built unit from Denon, with head amp and power supply on two separate chassis connected with an umbilical cord. The sound leaves a great deal to be desired, though. Much more edgy than that of the AU-340 transformer, it hardens and brightens to the point of nastiness on certain string passages and transients. We suspect feedback-related time modulation effects; not recommended.

Marcof PPA-1
(Improved)

Marcof Electronics, 7509 Big Bend Boulevard, Webster Groves, MO 63119. PPA-1 moving-coil pre-preamplifier, $124.95. Two-year warranty. Tested #006603, on loan from manufacturer.

Three new things have happened to our "Reference B" pre-preamp since the original review in the last issue. There has been a minor circuit change. The price has gone up five dollars. And the metal box is now black instead of blue. The net result is a considerable improvement in the sound, which of course was already excellent. (Must be the new color—or the extra money.) That slight trace of zinginess on top and looseness on the bottom are completely gone. The sound is now firmly controlled, beautifully focused and smooth as silk. The only thing that still separates the Marcof from SOTA quality is a lack of ultimate transparency and definition. Next to the Cotter transformer it sounds ever so slightly veiled and blunted, although its basic sonic character is very similar. At the price, nothing else comes even close. At twice the price, we still don't know of anything better. One of the great buys in audio today.

RWR Audio MCT-1

RWR Audio Ltd., Box 3080, Station D, 340 Laurier Avenue, Ottawa, Ont., Canada K1P 6H6. MCT-1 Moving-Coil Transformer, $299 (direct from manufacturer). Five-year warranty. Tested two preproduction samples, on loan from manufacturer.

Who would have thought it? A completely unknown little firm in Canada has come up with the only challenger to the supremacy of the Cotter transformer—at half the price and with six different switchable strapping configurations! In our opinion, the Cotter survives the challenge, but it remains on top by a very, very narrow margin. It took us long sessions of agonizing A-B comparisons to decide that the Cotter MK-2L (the best of all Cotter transformers so far) sounded just barely smoother and more neutral on top, with perhaps half a hairsbreadth more clarity, if that much. Nothing could have been closer except total identicalness.

And that's not all. The remaining minuscule difference may be due to quality control rather than design. We examined two RWR transformers, with a total of four channels. Each channel was slightly different in square wave response. The best channel had a small overshoot followed by quickly diminishing ripples, which is the correct response. The worst channel had an abrupt wrinkle right after the leading edge, indicating some kind of sharp glitch at around 130 kHz. The remaining two channels were between these extremes. Unfortunately, the best and the worst channel were in one transformer; the second best (which was almost right) and the third best (somewhat faulty) in the other. The two transformers didn't quite sound alike. We tried to listen to the best and second best channels in stereo but couldn't eliminate a hum caused by a ground loop between the two simultaneously used transformers. We have a feeling that a single transformer incorporating these two channels might have sounded identical to the Cotter. Our final conclusions were based on the slightly better-sounding of the two transformers, which was the one with the second and third best channels. The glitches are caused by something wrong in the windings, we believe; Cotter has such problems completely solved. We also suspect that the dip switches that select the strappings in the RWR are not as trouble-free as the less convenient soldered strapping connections used in the Cotter line.

Despite these minor problems, the RWR must be rated as a sensationally good product at a reasonable price, which at present is maintainable only by direct-to-user sales. More conventional distribution will undoubtedly raise the price in the end. We still consider the Cotter to be a more carefully made, better finished and marginally more accurate transformer, but the RWR is also very attractive in all these respects and still a lot better than any electronic MC step-up device known to us, regardless of price.

Signet MK12T

Signet Division, A.T.U.S., Inc., 33 Shiawassee Avenue, Fairlawn, OH 44313. MK12T Transformer for MC Cartridge, $300. Tested #934, on loan from dealer.

Another well-built Japanese transformer, with 3-ohm, 20-ohm and 40-ohm connections (plus bypass) switchable from the front panel, the MK12T doesn't quite make it to the top sonically. By itself it sounds reasonably transparent and neutral, but in comparison with the Cotter or even the RWR, which costs a dollar less, it has a somewhat hard and glassy quality. On dynamic peaks rich in high-frequency energy, the sound begins to approach nastiness but never really gets there. A possible though far from certain explanation is that the high-frequency resonance of the transformer is no higher than 54 kHz, which may be too close to the audio band for comfort. This is probably as good a transformer as most people are likely to hear in the course of their audio shopping in most places, but the very best it isn't.

Recommendations

For the first time, there's an obviously right choice at three different price points. As you know, we firmly believe that MC cartridges are the only way to go; therefore you should own one of the following in our opinion, no matter what your system consists of.

Best step-up device for moving-coil cartridges, regardless of price: Cotter MK-2 transformer.

Close to the best at a much lower price: RWR Audio MCT-1 transformer.

Best MC step-up device per dollar: Marcof PPA-1 pre-preamp (improved version).

Whatever Happened to the 'Admonitor'?

In response to comments from our subscribers, we're in the process of rethinking the format of the 'Admonitor' column, which is temporarily closed for alterations. When it reopens, the emphasis will be no longer on admonishing the hi-fi copywriter, whose forked tongue is apparently taken for granted even by novice audiophiles, but rather on monitoring the latest, most fashionable and most blatant audio hypes, which are a joint product of advertisers, dealers and reviewers. The column will probably end up as a kind of directory of Who's Kidding Whom About What.

Our First Tape Deck Review (We Picked a Good One)

Just a quick look at a remarkably well-engineered product, before we take up the subject in greater depth in a less crowded issue.

Tape recording and tape recorders are demanding to be explored with the same back-to-basics approach as we brought to the phono groove and tracking geometry, but we can't do it in this issue; it's much too big a subject. Just to get rid of typical audio salon notions of what's important and what isn't would take many more pages than we can spare here. The reeducative effort might have to be even more strenous than in the case of phono reproduction. Who talks today about absolutely fundamental concepts such as the relationship between tape speed and tape head geometry, the difference between amplitude modulation noise and phase modulation noise, or the essentially "digital" effects of high-frequency bias in analog recordings? There's a lot more to tape recorders than you can read even in the most complete spec sheet or manual.

Meanwhile, just to kick off the subject, here are our impressions of an excellent machine that caters to the well-heeled consumer rather than the professional.

Tandberg TD 20A

Tandberg of America, Inc., Labriola Court, Armonk, NY 10504. Model TD 20A stereo tape deck, 7½ and 3¾ IPS 4-track version, $1500. One-year warranty; manufacturer pays return freight. Tested #4503370, on loan from manufacturer.

When we borrowed this tape deck from Tandberg, the 15 and 7½ IPS 2-track version was not yet available. That's the one we really want to evaluate in depth, as it can be expected to utilize the inherent mechanical and electronic capabilities of the design to the utmost. Meanwhile, the slow-speed 4-track version under consideration here should be regarded more as an ultimate hi-fi tape recorder for the home than as a serious recordist's tool; however, it does demonstrate the extraordinary engineering skill Tandberg has developed over the years in making this type of deck. The performance of the machine exceeded our fondest hopes.

At 7½ IPS, we measured the peak-to-peak flutter and wow on a 3 kHz tone to be 0.027% for combined recording and playback. Assuming that these add up rms-wise, the transport itself checks out at 0.019%, a figure we find hard to believe. Man, that's stability—and you can hear it, too. Frequency modulation is the great killer of tape sound, but Tandberg has it licked. The published specs are obviously very conservative.

The frequency response of the TD 20A is, of course, partly dependent on recording level, as in other tape recorders, but at 7½ IPS it's essentially flat to 10 kHz over just about the whole dynamic range, with only 3 to 4 dB of roll-off at 20 kHz. We call that quite excellent. Even more impressive is the ability of the TD 20A to record and reproduce square waves with good leading and trailing edges and with decently flat tops. Most tape recorders flunk that test ignominiously. You have to push the Tandberg to a high recording level before the waveform begins to be non-rectangular. It's truly remarkable. Needless to say, all of these outstanding performance characteristics are somewhat impaired at 3¾ IPS, although the machine remains very, very good at that speed, far superior to the most pretentious cassette recorder.

As for actual music recording, we don't consider the 4-track format to be suitable for taping live concerts or studio sessions with top-notch microphones, not only because of the limitations in fidelity, which in the case of the TD 20A are highly tolerable, but also because of the impossibility of editing two-way tapes. We did, however, perform a very critical comparison between an outstanding direct-to-disc phonograph record as played on our "Reference A" system and a carefully made 7½ IPS copy of the record on the TD 20A through the same system. There was an audible difference, but our ears had to search for it very hard. Once we latched on to the sonic identities of the original and the dubbing, we could invariably identify them blind, but the spread between them was surprisingly small. The copy had a slightly less crisp and incisive top end and just a smidgen less inner detail. It was an outrageously good showing for a nonprofessional tape deck. In addition, our best prerecorded 7½ IPS 4-track tapes sounded better on the TD 20A than we had ever heard them before. What more can we say?

In conclusion, it should be pointed out that the controls and mechanical operation of the Tandberg are sheer delight. Our 15-hour seminar was recorded at 3¾ IPS on the TD 20A, and we could never have lasted through the rigors of editing the typed transcript against the original tape if the machine had been less smooth and responsive. Well done, Tandberg.

Cartridge/Arm/Turntable Briefs and Interim Reports

Our promised test of turntable resonances and acoustical breakthrough being delayed once again (for good and sufficient reasons but surely for the last time), we give a few unpostponable items the once-over, until more complete coverage in the next issue.

That's right. The turntable survey we announced two issues ago, with prcisely quantified comparisons of mechanical resonances and airborne excitations as measured in various designs, is still far from ready. We're shifting it to the next issue. Our timetable was way off, mainly because of the difficulty of putting a deadline on the solution of unsolved problems. (The last time *that* was done successfully was in World War II, on the Manhattan Project.)

The unsolved problem in this case was the selection of the proper test signal for energizing the turntable environment acoustically. We gradually came to the conclusion that white noise, pink noise, swept sine waves and other standard test signals used by other investigators known to us (see also page 55 of Vol. 2, No. 1) are all grossly unrepresentative of real-world listening room conditions. Actual music, of course, is the most authentic test signal, but it's nonrepetitive and very difficult to extract any kind of steady meter reading or CRT display from. We believe we now have the right test signal, one that models the spectral energy of music correctly and can still be easily dealt with on the lab bench; the rationale will be explained in the next issue in conjunction with the test results. Chalk up this delay to our reluctance to be wrong. We'd rather be late but right.

Meanwhile we're giving below our capsule views on a number of interesting items, so as not to keep you in suspense for another few months. The criteria on which these views are based were explained at some length in the last four issues; here we shall restrict ourselves to the conclusions, which may very well be revised (though in no foreseeable case entirely reversed) after the accumulation of more complete data.

A word about cartridges.

We're also in the process of implementing a realistic laboratory measurement program for phono cartridges, the early results of which will be visible in the next issue. Until then we want to warn you about a totally invalid approach that has only recently emerged from the witch doctor's hut and is beginning to be taken seriously by a few wide-eyed audiophiles. It consists of dropping the stylus of a mounted cartridge on a hard surface, such as glass, and spectrum analyzing the resulting electrical impulse on a Fourier analyzer. The spectral composition of the impulse is claimed to correlate with the audible performance of the cartridge. Since a pickup reproduces music by tracing the groove walls with an indentor, any test method that totally ignores the physics of the stylus/groove interface is of monumental naivete. Should you ever encounter any cartridge recommendation based on this test, our recommendation is that you start walking rapidly in the opposite direction.

Speaking of the stylus/groove interface, we've been recently exposed to an extremely interesting new experimental stylus design. The diamond tip takes the line-contact idea to its logical limit; the contact area has by far the largest aspect ratio (length to width) ever attempted, extending over almost the entire height of the groove wall. This of course mimics the cutter stylus geometry to the utmost. The stylus beam in which this diamond was mounted had been specially deadened to make it as inactive in terms of energy storage as possible; the cartridge body was that of a standard Japanese low-impedance moving-coil unit—it doesn't really matter which. The whole thing was a preliminary study for a totally new moving-coil cartridge, which should be out sometime in 1980 and which we were asked, as the price of being privy to this early experiment, not to identify by name. (Please don't write us or call us.) The new cartridge will have a similar stylus, a considerably more sophisticated dead-beam cantilever, and a totally new generator mechanism, unrelated to that of the experimental cartridge we played with. The projected price is $350, but projected prices have a way of going up.

The point is that this rather primitive experimental exercise already sounded better than the Koetsu, which is today's State-of-the-Art moving-coil cartridge! The highs were considerably cleaner and more natural; the midrange was comparable though no better; the output was lower but will be considerably higher, we're told, in the new cartridge. Overall, the Koetsu sounded slightly "electronic" next to this makeshift hybrid. The designer claims that when the whole act is together and the finished cartridge is released, all other cartridges will become Paleolithic artifacts overnight. We're always skeptical of such predictions, but in this case we must say there just may be some basis to it. As in loudspeakers, there's a lot of room for progress in phono cartridges. What we now have is far from perfect.

D B Systems DBP-10 Protractor

D B Systems, PO Box 187, Jaffrey Center, NH 03454, DBP-10 Phono Alignment Protractor, $19.95.

This isn't much more than a very durable and precise version of the homemade cartridge/arm alignment protractor we described on page 47 of Vol. 1, No. 6. It's made of thick, tough plastic; it has various useful auxiliary lines and grids printed on both sides; in addition, it comes with a transparent plastic overlay printed with markings that permit you to hunt for the correct overhang without having to swing the cartridge all the way over the spindle. The principle of the DBP-10 is that if the cartridge is in perfect alignment at both of the two zero-error radii, then the overhang is automatically correct. There's nothing wrong with this approach, except that we don't find it a hardship to measure the overhang directly. Of course, there *are* some situations where it's impossible to swing the arm further in than the lead-out grooves. In any case, this is a very neat and convenient way to have a permanent alignment protractor, correctly calibrated. Highly recommended.

Dennesen Geometric Soundtracktor

Dennesen Electronics, PO Box 51, Beverly, MA 01915. Geometric Soundtracktor, $35 in plastic, $100 in aluminum.

A more ambitious and elaborate gadget than the above, the Dennesen makes use of a clever geometrical insight to make possible correct lateral tracking alignment in one shot, with a single adjustment.

The design is based on the fact that, when the alignment is correct for standard LP records, the product of the effective arm length and the sine of the offset angle is always 93.4 mm, regardless of the arm used. If you set up a right triangle such that its hypotenuse is the line from the arm pivot to the inner zero-error point, at a radius of 66.0 mm, and its shorter side is an extension of that radius through and past the turntable spindle by an additional 27.4 mm for a total of 93.4 mm (66.0 + 27.4 = 93.4), then your lateral tracking geometry is locked in; all you need is perfect tangential alignment at that one zero-error point. The Dennesen is a precision tool that enables you to do this very conveniently, in a matter of minutes—but there's a hitch. If you don't locate the lateral pivot point of the arm right on the button, all bets are off. The final alignment could then be grossly inaccurate in certain cases, even if you performed all subsequent steps perfectly. This isn't just nit-picking, since the exact pivot point is often very hard to determine from the top. It would have been much better to extend the base plate of the tool another couple of inches in order to include the outer zero-error point as a verification check. There's safety in redundancy. (The extra accessory Dennesen does give you, instead, is a rather awkward VTA indexer, which is supposed to help you come back repeatedly to the correct VTA for a particular record, once you have found it by ear. All we can say about it is that it's primitive but harmless.)

Incidentally, the "pat. pending" legend on the Geometric Soundtrack raised some eyebrows in our circles. Is Dennesen trying to patent a mathematical truth? It won't be a very strong patent in that case, even if the patent examiner doesn't quite understand the subject and agrees to everything. As a well-made and handy tool, however, the device has our complete approval—as long as you know *exactly* where the arm pivot is located.

Denon DA-401

American Audioport, Inc., 1407 North Providence Road, Columbia, MO 65201. Denon DA-401 Integrated Tone Arm, $360. Tested sample on loan from importer.

A handsome and beautifully made carbon-fiber tone arm, the DA-401 has slightly wobbly bearings and no provision for adjusting the VTA during play. At this stage of the game, that's not what is expected of a $360 arm. Too bad, especially since the geometry is almost 100% correct (though not the overhang instructions) and the antiskating mechanism particularly nice.

Denon DL-303
(interim report)

American Audioport, Inc., 1407 North Providence Road, Columbia, MO 65201. Denon DL-303 Moving Coil Cartridge, $385. Tested sample on loan from importer.

Our initial impression of the new top-of-the-line Denon cartridge is that it isn't as good as their older DL-103D, which in turn is several small steps below our current recommended choices. See final review in the next issue.

Denon DP-80

American Audioport, Inc., 1407 North Providence Road, Columbia, MO 65201. Denon DP-80 Quartz Locked Turntable, $960 (complete with required voltage step-up transformer). Base, $690. Tested sample (without base) on loan from importer.

This is the top-of-the-line Denon with the big motor we briefly referred to on page 59 of the last issue, in anticipated comparison with the Technics SP-10 Mk II. Well, now we know that the DP-80 doesn't have quite as much torque as the SP-10 Mk II, but in the Cotter B-1 base it certainly works just as well and sounds just as good. We still consider the magnetic tachometer system of the Denon direct-drive turntables to be a bit fragile and fussy, though certainly superbly accurate, and we still like the brute-force brake system of the SP-10 Mk II better; on the other hand, the latter seems to be much more difficult to obtain these days and the Denon isn't in any way inferior in performance. The DP-80 has no third speed, though, and on U.S. voltages it only works with the accessory step-up transformer. (Not enough demand for a separate U.S. model.) Our verdict for the moment: SP-10 Mk II for anvil-like utility; DP-80 for electronic sophistication; either one in the Cotter B-1 base for the ultimate in sonic accuracy. The factory bases are another matter altogether.

DiscFoot Isolation System
(interim report)

Discwasher, Inc., 1407 North Providence Road, Columbia, MO 65201. DiscFoot Turntable Isolation System, $22 (four feet). Additional feet, $5.75 each. Tested samples on loan from manufacturer.

We've made fun in the past of the little rubber nipples and ineffectual pillboxes most turntables stand on. Well, the DiscFoot is better, though not all that different in physical appearance. Just how much better will become clear when we complete our turntable resonance tests; you can expect a full report in the next issue.

Meanwhile, we can tell you that the DiscFoot system won't appreciably lower the resonant frequency of the turntable suspension, so it can't provide additional isolation at the lowest frequencies. Nor can it protect in any way against airborne excitation. What it can do is to lower the Q of system resonances in a somewhat higher frequency range, thereby isolating certain kinds of mechanical feedback and footfall vibrations. It does this as a result of the chemical/physical properties of the material it's made of. Very interesting and definitely deserving of further attention.

Fidelity Research FR-14

Fidelity Research of America, PO Box 5242, Ventura, CA 93003. FR-14 Precision Tone Arm, $400. Tested #011187, on loan from distributor.

This is an arm of the same length as the FR-64s and made with the same tender loving care. Everything about it is simply gorgeous, including the bearings. It isn't dynamically balanced, however, like the FR-64s, nor does it have provisions for VTA adjustment during play. At $400, that begins to look like a not particularly good buy in this age of VTA enlightenment, especially in comparison with the much cheaper JVC UA-7045 (see below).

JVC MC-1
(interim report)

US JVC Corp., 58-75 Queens Midtown Expressway, Maspeth, NY 11378. MC-1 Direct Couple Type Moving Coil Cartridge, $300. Tested #10300174, on loan from manufacturer.

The final, full production version of this cartridge has none of the irritating high-frequency coloration we reported in our review of the prototype on page 45 of Vol. 1, No. 5. The highs are smooth and pleasant; the overall sound is very clean and transparent. If it weren't for the $230 Fidelity Research FR-1 Mk 3F, we wouldn't hesitate to rate the MC-1 as by far the finest moving-coil cartridge in its price range. The FR, however, appears to us to have even greater midrange clarity, approaching that of the Koetsu, and also a more completely neutral character. This may not be a final verdict, since our exposure to the JVC was somewhat limited. See the next issue for a more thorough evaluation.

JVC TT-101
(interim report)

US JVC Corp., 58-75 Queens Midtown Expressway, Maspeth, NY 11378. TT-101 Direct Drive Turntable, $1000 (with base).

JVC's answer to the Denon DP-80 and the Technics SP-10 Mk II came to us so close to press time that we couldn't even set it up and listen to it, let alone test it for real. We do want to tell you, however, that the construction and craftsmanship are of the highest order, as in other top-of-the-line JVC 'Laboratory' products. Big motor, lots of torque (though not quite as much as in the SP-10 Mk II), two speeds, very elegant speed control with 13 discrete and repeatable steps—these are the immediately apparent features even before actual use. We see nothing in this unit that would obviously prevent it from being the best of them all, but only the full test will tell. The factory base, on the other hand, is totally pedestrian; the Cotter treatment is definitely called for.

JVC UA-7045

US JVC Corp., 58-75 Queens Midtown Expressway, Maspeth, NY 11378. UA-7045 Tone Arm, $250. Tested #12405142, on loan from manufacturer.

We're simply delighted with this well-designed arm, which looks to us like the nearest thing to a Fidelity Research FR-64s, at a fraction of the price. Maybe it isn't quite as beautifully constructed but it's close, and it's extremely convenient to set up and to use. It lacks dynamic balancing, to be sure, but the bearings are nice and tight, the arm tube doesn't ring, the antiskating bias adjustment is a cinch, and—surprise, surprise!—the VTA is tunable during play. All that and a very dead magnesium headshell, too—what more can you ask for $250 in this day and age? We'll call this one Reference A Minus.

Kenwood KD-650
(interim report)

Kenwood Electronics, Inc., 1315 East Watsoncenter Road, Carson, CA 90745. KD-650 Direct-Drive Turntable, $400 (with arm). One-year warranty. Tested #960133, on loan from manufacturer.

This is an improved version of the Kenwood KD-500 turntable that was our original "Reference B" recommendation. The main improvement is the quartz-locked direct drive; the "resin concrete" chassis is similar and, though not totally inactive acoustically, a lot heavier and deader than the resonant tin cans you generally get at this price; the workmanship and finish appear to be better; and the tone arm that comes with the KD-650 version (the armless version, KD-600, costs $50 less) is surprisingly sophisticated in design, with very convenient though limited-range VTA adjustment during play. How about that? We detected a very small amount of wobbly "give" in this arm, but it doesn't seem to come from the bearings. The probable culprit is the setscrew holding the arm pillar in the sleeve; this is a curable disorder that will be dealt with in our final review.

The main fault of the KD-650 is once again that it has no suspension to speak of, relying almost enitrely on its inertial mass for isolation. A Cotter B-2 isolation platform would be a great addition, making it a $595 system complete with arm, which is still within "Reference B" range, though just barely. Even so, we don't know of anything better for the money.

Linn-Sondek LP12
(reappraisal)

Linn Products Ltd., Glasgow, Scotland; distributed in the USA by Audiophile Systems, 5750 Rymark Court, Indianapolis, IN 46250. Linn-Sondek LP12 Transcription Turntable, $865. Two-year warranty (electrical components one year). Tested #019794, on loan from distributor.

Our candid avowal that the main bearing of our original test sample hadn't been filled with oil (see Vol. 1, No. 6, p. 56) stirred up a full-scale tempest in the small but steamy teapot of Linn-Sondek groupies. There's something about an audiophile-oriented product having some technical merit but also some very basic flaws (Magneplanar is another example) that engenders a highly defensive cultism among its sellers and buyers. They'll tell you that if you're experiencing the flaws, you're doing something terribly wrong—because the product is well-nigh flawless. Doing everything right involves a whole series of rituals, all of them having to do with second-order and third-order effects and designed to draw your attention away from the first-order defects. For example, the U.S. distributor of the Linn-Sondek told us that we mustn't mount our cartridge with anything but steel screws and nuts because other metals don't provide enough rigidity and alter the sound. What he didn't tell us, on the other hand and among other things, was that the Linn-Sondek's tinny little subchassis holding the platter, main bearing and arm board is acoustically so live that you can twang it like a banjo. Some perspective! No wonder this crowd went into a paroxysm of I-caught-you-red-handed when they read about the bearing oil.

Now anyone who isn't completely untutored in the fundamental technical disciplines would grasp immediately that the amount of oil in the bearing well isn't a vital, all-or-nothing link in the complex mechancial system that couples the stylus to the groove in motion. It represents a teeny-weeny fraction of the total slack and wobble in the system, unlikely to have an influence on audible effects significantly above threshold. That indeed turned out to be the case when we retested the turntable; the consensus was that the difference was either inaudible or so small as to be uninteresting even to the purist. Of course, once the vial of oil was added, it was impossible to A-B back and forth between the two conditions for further verification. It should also be remembered that there's always quite a bit of lubrication in the bearing well as the turntable is delivered from the factory; we never tested a totally dry condition. The whole thing was an exercise in techno-fetishism.

In fact, our reexamination of the Linn-Sondek actually lowered our opinion of it in a number of respects. We've already mentioned the liveness of the mounting plate; both the Cotter and the Win systems are incomparably deader. Another problem is that the drive belt can be sufficiently displaced by violent low-frequency excitation of the suspended system to cause audible modulation of the sound. All you need is a piano record with thunderous bass, a pair of really good subwoofers and a fairly resonant floor. Compared with the Cotter B-1, for example, the Linn-Sondek will occasionally sound a little dithery under these circumstances. There's no substitute for high mechanical impedance, i. e., large inertial mass. (See also our original review in Vol. 1, No. 5 for the various disadvantages of small mass *cum* soft springs *cum* high Q.) Add to all this the lack of 45 RPM, lack of speed control, lack of first-rate appearance and finish except for the moving parts, and the new price of $865 will begin to look like a rip-off. What it buys is, after all, not much more than a glorified AR turntable.

A few users have tried to improve the Linn-Sondek by immobilizing its suspension, putting it on a Cotter B-2 isolation platform, and burying all undeadened structures in sight in gobs of Duxseal. At that point you've got one hell of an ugly $1060 turntable that's a lot less active mechanically and acoustically. You've also got a whole new set of enemies, including the manufacturer, the distributor and most dealers of the Linn-Sondek. Our impression is that these people can't deal with technical criticism and scientifically enlightened debate; their response is a mystical assertion of unarguable superiority and a pitying condescension toward incompetents holding contrary opinions. You can experience it at shows and read about it in various printed bulletins. The Linn-Sondek is the theocratic turntable.

Ortofon MC30
(interim report)

Ortofon, 122 Dupont Street, Plainview, NY 11803. MC30 Moving Coil Cartridge, $600. Tested #841895, on loan from manufacturer.

Our biggest disappointment in cartridges so far, especially after the tantalizing comments made about it in our State of the Art seminar (see Vol. 2, No. 1, p. 33). The initial listening tests gave evidence of considerably less clarity and more noticeable colorations than we heard with much less costly MC cartridges in our reference system, such as the FR-1 Mk 3F, the JVC MC-1 and the GAS 'Sleeping Beauty' Shibata. We seriously doubt that this conclusion will be reversed by our more probing investigations to be reported in the next issue, but we sure hope so. This is no $600 cartridge the way we see it now.

Thorens TD 115
(interim report)

Elpa Marketing Industries, Inc., Thorens and Atlantic Avenues, New Hyde Park, NY 11040. Thorens TD 115 Semiautomatic Turntable, $450 (with integral arm). One-year warranty. Tested #12477, on loan from manufacturer.

We expected "Reference B" quality from Thorens at this price, but no such luck. The chassis is of the resonant tin-can school; the suspension is of highly questionable design, with insufficient travel; the entire structure is light and flimsy; and—horror of horrors—the arm is shock-mounted in such a way that the tracking geometry can be modulated during play. The whole concept reflects a total disregard of the basic physics of phono reproduction and a preoccupation with nonessentials. Maybe our quantitative tests will mollify this harsh judgment, but our ears certainly don't. Where's that old Thorens know-how?

Wheaton 240 Type II

Wheaton Music, Inc., 2503 Ennalls Avenue, Wheaton, MD 20902. Model 240 Type II Decoupled Precision Tone Arm, price NA. Tested sample on loan from manufacturer.

Herbert Papier, an affable gentleman who comes from the watchmaking discipline and is now the proprietor of Wheaton Music, has been after us for almost two years to say something about his handcrafted tone arms. They're very precisely made, with a great deal of attention to mechanical detail, though not necessarily with a total systems view of the phono arts.

The first Wheaton arm we saw was a Rabco SL-8E modification, incorporating some very sensible mechanical improvements; it had a hollow balsawood arm, however, with a characteristic midrange coloration that could be evoked by tapping and was also audible in the music. Regardless of that, we don't consider the basic Rabco configuration, which requires virtual disassembly for height adjustment, to be a viable arm in this new era of tunable VTA.

The current Wheaton 240 Type II arm is a pivoted design of correct geometry and a larger range of VTA adjustment during play than is available in any other arm known to us. The design comes off as a bit Rube Goldberg-ish, with too many parts sticking out in too many places, but they're all beautifully machined and work very well. Among other things, the antiskating bias can be disabled during cueing—a very nice touch. Again, the Achilles' heel of the arm is the same square-cross-section, hollow balsawood design of the arm tube, which is much too live and has the same sonic signature when tapped—and, of course, when playing the record. Obviously, the lighter-is-better fallacy has been working overtime here. A more useful precept for guiding Mr. Papier's skilled hand would be that the only good tone arm is a dead tone arm.

Recommendations

Things seem to be moving more slowly in this product category than in others, as the familiar faces below will attest.

Best phono cartridge in actual production, regardless of price: Koetsu.

Close to the best at a very much lower price: Fidelity Research FR-1 Mk 3F.

Best tone arm, regardless of price: Fidelity Research FR-66s (if you have the room for it) or Fidelity Research FR-64s with B-60 stabilizer.

Close to the best at a very much lower price: JVC UA-7045.

Best turntable, regardless of price: Cotter B-1 system with specially adapted Technics SP-10 Mk II or Denon DP-80.

Best turntable/arm per dollar: Kenwood KD-650.

FM Tuners: A Hopeless Dilemma for the Serious Audio Reviewer

A dependable evaluation of FM tuners from the audio purist point of view appears to us at the moment to be both unfeasible and irrelevant. Nevertheless, we report what little we've found out about just a few interesting tuners.

The central fact of the present-day FM scene is that nearly all stations are broadcasting a signal of inherently poor audio quality, not much more enjoyable when received with an excellent tuner than over the kitchen radio. Most of the music is on beat-up LP records, played with inferior pickups misaligned both laterally and vertically; the occasional live broadcast is likely to be carelessly microphoned with second-rate equipment. So the first and most obvious question is—why would anyone want to buy an expensive FM tuner of audiophile caliber to listen to this junk?

The best answer we can expect is that a few metropolitan areas in the country—maybe three or four—each have *one* good-music station that worries about audio quality, but even that isn't a good enough answer. We have perhaps the best such station, WNCN New York, about 18 miles from our antenna, and we can tell you that the *same* records sound incomparably better when played on our "Reference A" system than when received from WNCN via our Sequerra Model 1 tuner plugged into the same system. Why? Because no FM station, not even the fussiest, uses top-notch moving-coil cartridges with line-contract styli (not rugged enough for back-cueing by DJ's), and no FM station tunes the VTA separately for each record, even if the tone arm is correctly aligned for lateral tracking geometry.

We don't consider the sound of WNCN to be sufficiently transparent and focused to serve as a reference in determining the limits of a tuner's ability to resolve audio information and in making meticulous A-B listening comparisons. Not that it isn't a good enough sound to permit some tentative conclusions (see the brief reviews below), but the confidence level just isn't the same as in our preamp comparisons, for example. We hardly need to add at this point that, without such listening tests, all bets are off; tuners are subject to the same elusive time-domain distortions as other electronic devices in the signal path, and you already know how far black-box measurements on the lab bench get you in looking for these things. So, just for openers, FM tuner quality is both academic and inscrutable, at least under the present circumstances.

It has been suggested that a reference-quality audio signal, out of a preamp or a tape deck, could be fed via an FM signal generator directly into the antenna terminals of a tuner under evaluation. That would certainly eliminate any possible objection to the program source; unfortunately, real-world conditions aren't duplicated by such a listening test. One of the most important requirements in an FM tuner is that it mustn't act as an AM tuner. All AM, including multipath, must be rejected as sharply as possible, otherwise it will confuse the FM detector and impair the sound. This crucial aspect of clean FM reception isn't tested at all when the front end of the tuner is driven from an FM generator. The winner in this kind of listening comparison could conceivably be a loser when the roof antenna is connected. Incidentally, one of the most common shortcomings of routine FM tuner testing on the lab bench is the lack of attention to AM rejection. Sometimes it can be the whole ball game.

Maybe what we should do is to rent a local FM station with a good transmitter during the night hours when it normally doesn't broadcast, drag the phono and tape components of our reference system there, and run a listening comparison of tuners in our laboratory. That would probably work. Frankly, we don't think it would be worth the effort, even if they allowed us to do it. We'll wait until the average quality of FM broadcasting improves a bit. Meanwhile, here are our impressions of a few tuners, old and new, on the basis of A-B listening comparisons using WNCN and other local stations as the program source. We doubt very much that we'll do this again in the near future.

NAD 4080

NAD (USA), Inc., New Acoustic Dimension, Mackintosh Lane, PO Box 529, Lincoln, MA 01773. Model 4080 AM/FM Stereophonic Tuner, $285. Two-year warranty. Tested #4806152, on loan from manufacturer.

This is from the same people who make the astonishingly good low-priced integrated amplifiers reviewed elsewhere in this issue. The 4080 is their top-of-the-line tuner (yes, the others cost even less); like all NAD products it's made in Taiwan, of unimpressive parts that are nevertheless of reasonably good quality in strategic places. The construction and cosmetics are well above expectation in this price category. Lots of features, too, including multipath indicator and switchable Dolby equalization (though no decoder).

In direct A-B comparison with our Sequerra Model 1 reference tuner, the NAD sounds a little less transparent, with inner detail not quite as airy and sharply defined. It's also a bit noisier. All in all, however, the difference is amazingly small, even to the point of inaudibility on some program material. We don't see why anyone would need a better tuner than this, except possibly to tape record the one or two lives broadcasts a year that are genuinely clean. The 4080, like most good tuners, is still heavily overqualified for the job of faithfully reproducing day-to-day FM garbage.

Sequerra Model 1

The Sequerra Company, Inc., 143-11 Archer Avenue, Woodside, NY 11435. Model 1 FM Tuner, $3600. Five-year warranty. Tested #1022, owned by The Audio Critic.

Still the acknowledged *ne plus ultra* of FM tuners after all these years, despite its lack of quartz-locked tuning and other newfangled wrinkles, the Sequerra Model 1 survived Dick Sequerra's departure from the company bearing his name and is made today pretty much the same way as ever, except perhaps without Dick's fanatical attention to alignment.

A full description of the tuner would take pages and is probably unnecessary in view of its long-standing reputation; we still chuckle, though, whenever we use its "panoramic display," which is actually an RF spectrum analyzer. The legend is that this started as a joke; somebody said to Dick, "Come on, what can you possibly put into a tuner for that kind of money—a spectrum analyzer?" and Dick said, "Yes!" Then he just had to do it and figure out some justification for it. But he also put all sorts of other good things into the tuner, and it all works very nicely. We own the 22nd one ever made, and it has never given us any trouble.

Sonically the Sequerra is a small cut above any other FM tuner known to us; it's just a little cleaner, more transparent, better focused, more detailed on top, and quieter than the others. But the difference isn't dramatic; it's audible only on truly exceptional broadcasts. In fact, most of the people we know who buy Sequerra tuners aren't typical audiophiles but obsessive elitist perfectionists who can't stand the thought of anything but the best. Like us.

Series 20 Model F-26

Series 20 (a division of Pioneer Electronic Corp.), 85 Oxford Drive, Moonachie, NJ 07074. Model F-26 FM Tuner, $1000. Two-year warranty. Tested #YF3600195M, on loan from manufacturer.

Very much in the contemporary Japanese high-end tuner idiom, the F-26 is beautifully built, elegantly functional, highly automated, with quartz-locked tuning and all that jazz. In A-B listening comparisons, however, it doesn't quite measure up to the Sequerra Model 1, sounding just a bit more constricted overall and less wide-range (or call it less fast and defined on top). The Yamaha CT-7000 also appears to be marginally superior to it in these respects. We can't get too excited about the differences, though; the F-26 is still a very fine tuner in a world of lousy FM stations.

One feature that gave us a little trouble is the automatic (and undefeatable) switching between wide and narrow IF passband, designed to track the incoming signal quality as sensed by the tuner and provide optimum audio quality at all times. The circuit can occasionally be fooled by irrelevant phenomena such as, for example, the presence of an SCA subcarrier. The switching between the two IF bandwidths is accompanied by a click, which can ruin tape recordings, among other things, if it comes in the middle of the music. This erratic behavior is probably the only negative aspect of what is otherwise a very slick little FM tuner.

Yamaha CT-7000

Yamaha International Corp., 6600 Orangethorpe Avenue, Buena Park, CA 90620. Model CT-7000 FM Tuner, $1200 (price as of late 1977, unit no longer made). Tested #1735, on loan from owner.

When this tuner first appeared on the scene, it was hailed as the only possible rival of the Sequerra Model 1, at a fraction of the price. Some even claimed it was indistinguishable from the Sequerra in audio quality and a number of other respects; a few said it was actually better. Our A-B comparison indicated a JND (just noticeable difference) in favor of the Sequerra, but as we pointed out these FM listening tests aren't very rigorous. The Sequerra appeared to have just a shade better top-end definition, transparency and dimensionality, but it was by far the smallest and most elusive difference in any of our FM tests. On indifferent program material the two tuners sounded absolutely identical.

Some time ago, the CT-7000 was discontinued, but there are always a few of them being traded in the secondhand market. At a good discount, we'd say it's an unbeatable bargain. Meanwhile, the Sequerra is hanging in there as the sole representative of unmitigated overkill in FM tuners, and just one of those is more than enough in our opinion.

Recommendations

None. Why would you want to buy expensive new shoes when all the streets in town are unpaved and muddy? Use whatever you have or whatever you can get cheap. Or pick one of those reviewed above, if it intrigues you. When it comes to FM the way it exists in the U.S.A. (quite aside from its inherent potential), the reaction of **The Audio Critic** is to walk away from it.

Yes, Preamplifiers Are Still Getting Better All the Time

By the Staff of
The Audio Critic

No new SOTA ratings this time, but some of the lower-priced preamps are beginning to sound so much like the best that maybe everybody has been slowly but surely moving in the right direction after all.

We're still putting all preamps through the various "black box" bench tests discussed in our earlier issues, but it's becoming quite clear that such tests reveal only obvious design errors and specific malfunctions, rather than the engineering subtleties that separate good, better and best. (See also our comments on power amplifiers in this regard.) Elusive dynamic distortions are impossible to read out directly on meters and CRT's; one must go by indirect, inferential routes from possible symptoms to likely conclusions. It isn't an exact science, and there's still no substitute for the long-suffering listening comparison by insertion into a known reference system of high resolution.

High resolution—there's the rub. You don't get it by using a Shure V15 Type IV cartridge, which is electrically a 13-kHz low-pass filter network; you don't get it by neglecting the lateral and vertical tracking angle alignments, which can result in as much as 5% FM distortion; you don't get it by A-B-ing everything through AR speakers, which ignore the time domain as a matter of fundamental design philosophy. How many preamp reviews have you read that were based on listening tests implemented with perfectly aligned moving-coil cartridges having line-contact styli, with properly isolated and deadened turntables, with electrostatic speaker systems—need we go on? No wonder that all preamps sound the same to so many reviewers; don't all furs feel the same through wool mittens and don't all wines taste the same through spearmint chewing gum? The preamps reviewed below were inserted into our Reference A system (see updates elsewhere in this issue), and the sonic differences we perceived may quite possibly not be resolved by the demo setup in your dealer's showroom.

This time some of these differences were surprisingly small, but not for the reasons that would be assumed by the all-preamps-sound-alike faction. (Yes, we do measure the RIAA equalization to the nearest 0.1 dB; yes, we do match volume levels when we A-B. Try again.) The principal reason is that when two designers working independently both understand what the problem is, they may come up with different solutions, but there will be a convergence toward functional truth, i.e., correct performance. Two 100% correctly performing preamplifiers *should* sound indistinguishable from each other, no matter how different they are in circuit design; it isn't quite happening yet, but the differences in some cases have been getting smaller and smaller. We still haven't found anything to equal the **Cotter System 2** in transparency, dynamic range, resolution of detail and day-to-day consistency of performance; unfortunately, the total price of the five modules is now a staggering $3280, of which the three that correspond to other "complete" preamps add up to $2230. Of these three, the CU-2 Control Unit, at $1450, doesn't really exist yet, the dozen or so physically uncouth "engineering models" made for a few dealers and professional users being the only ones around. The production version will be coming soon, we're told.

The encouraging thing, however, is that a number of far less expensive preamps give *almost* as good sonic results as the Cotter. This was not the case until very recently. Whether these lower-priced preamps will sound as good after five years of use as they do when they're new is another matter altogether. Ruggedness of construction and quality of parts still appear to be pretty much in proportion to price. It's only intelligent circuit design that has become cheaper.

Audionics RS-1

Audionics, Inc., Suite 160, 10950 SW 5th Avenue, Beaverton, OR 97005. RS-1 Preamplifier, $699. Three-year warranty. Tested #25030, on loan from manufacturer, and #25072, on loan from dealer.

This highly appealing unit—well built, versatile and nice to handle—is imbued with the kind of engineering integrity and basic good sense that Audionics stands for in our book. Sonically it also rates high; it must surely be one of the top six or seven preamps we've listened to so far, regardless of price. Just a year or two ago, we would have raved about this degree of clarity and neutrality; today, however, the RS-1 represents neither the best sound there is nor even the best sound for the money. The best combination of reliable circuitry, quality construction, features *and* sound for the money—maybe.

What the RS-1 lacks in audio quality, and what even some of the less expensive new preamps approximate more closely, is that ultimate degree of transparency and delicacy of detail, combined with a total absence of nasal colorations, high or low. Not that the RS-1 isn't quite excellent in all these respects; it just doesn't make it all the way to the finals. The earlier of our two samples also had a very slightly hard or strained quality on some kinds of program material; the second sample sounded quite sweet and smooth.

Threshold-level imperfections like these won't show up on the laboratory bench; just in case anyone should suspect the RIAA equalization, it's right on the button below 1 kHz and rolls off just a hair more than it should at the higher frequencies, the error being -0.35 to -0.45 dB at 20 kHz, depending on the sample measured.

Many users will appreciate the excellent tape-copying facilities; others will be intrigued by the "axial tilt correction" feature, which is a somewhat oversimplified attempt to correct electronically for the least important misalignment of the pickup stylus, namely lack of perfect perpendicularity as viewed from head-on. A test record is supplied to calibrate the adjustment; the whole thing is a nice little extra touch that shouldn't influence the audio purist's buying decision one way or the other. All in all, a very honorable effort.

Audire 'Legato'

Audire, Inc., 9576 El Tambor Avenue, Fountain Valley, CA 92708. 'Legato' Preamplifier, $330. Three-year warranty. Tested #4071, on loan from manufacturer.

We're tempted to praise this cute little pancake-style (only 1¾" thick) preamp for its surprisingly good sound at a price where we didn't expect much, but we realize that we never *really* heard what it sounds like. It has a kind of built-in tone control in the form of grossly inaccurate RIAA equalization.

The RIAA error curve of the preamp fills out a strip 1.7 dB wide. You could cut this in half and say that the error is ±0.85 dB, but that would be misleading. Most of the error is minus, in the form of a giant saddle starting at 1 kHz, dipping to approximately -1.25 dB in the 8 to 9 kHz area, and curving back to the zero line at the ultrasonic frequencies. That gets rid of a lot of high-frequency energy right in the region where an overbright or edgy quality will intrude if the circuit suffers from certain common design flaws. The Audire unit just barely flirts with such high-frequency stress on program material that readily provokes it; to our ears, it appears to be better in this respect than many far costlier preamps. But what if the RIAA equalization were restored to normal? Would there be a lot more zip and hardness? We have no idea. There would be a slightly different tonal balance, that's for sure.

So all we can tell you is that the Audire 'Legato' is well constructed considering its price, has a nice open sound without attaining the ultimate in transparency and inner detail, doesn't seem to have any obtrusive colorations, and would probably sound somewhat different with accurate phono equalization. That eliminates it from audio purist consideration, but a corrected version should be very interesting.

Bauman PRO-400

Bauman Research Instruments, Inc., Route 1, Box 52UA, Rosenberg, TX 77471. PRO-400 Stereo Preamp, $1050. Five-year warranty. Tested early production sample, on loan from manufacturer.

This is a very ambitious high-end audiophile product; just how ambitious should be apparent from the manufacturer's announcement that later production runs will be converted to Camac connectors, a la Mark Levinson. That $1050 price tag also has a Levinsonesque *déjà vu* about it; that was the price of the old JC-2, before the ML-prefixed, silver-wired Saudi Arabian status symbols spiraled up to the multi-kilo-petro-buck bracket. The Bauman, however, also offers a switchable moving-coil head amplifier plus tone controls as part of the package.

Interestingly enough, the Bauman PRO-400 sounds very much like an early ML-1 with a little extra zip. In other words, very open, very clean, very quiet, but rather hard, bright and fatiguing. In addition, we discern in the Bauman a characteristic midrange coloration of its own. Frankly, it isn't an acceptable sound to the perfectionist ear.

On the test bench, the PRO-400 performed beautifully. In fact, our impression is that it was designed to perform beautifully on the test bench, rather than in the ear of the beholder. The spec sheet and the promotional literature abound in very small and very large numbers; any reference to music appears to be deliberately avoided. We suspect the presence of some of the elusive feedback-related problems discussed at length in our seminar, but we made no heroic effort to track them down. On all routine tests the PRO-400 comes through with flying colors, confirming the spec sheet in every respect, including those beautiful square wave photographs. The RIAA equalization error is nil above 1 kHz; at the lower frequencies there's a broad saddle dipping to -0.4 dB in the 100 to 200 Hz octave, an anomaly predictable from the topology used but still within specs.

Physically the unit is very well made; its switching and control facilities are quite complete; we wish we were able to be more enthusiastic about it, but we know we could never live with that overbright sound.

Hegeman 'Hapi Two'

Hegeman Audio Products, Inc. (Hapi), 176 Linden Avenue, Glen Ridge, NJ 07028. Hapi Two preamp/control unit. $900. Two-year warranty. Tested #600, on loan from manufacturer.

There isn't really anything new to report about the Hegeman preamp circuit, which remains unchanged in this new and very stylish extra-flat package with brushed-aluminum relay-rack panel. We just want to make it official that we've tested it again and it's as good as ever, perhaps even a little better in the latest production runs, as a result of more completely systematized manufacturing and quality-control procedures. It sounds simply beautiful.

From phono input to main output, the only preamp known to us with even greater transparency and resolution of detail is the more than twice as costly Cotter, the control module of which isn't really in production yet. Comparing phono sections only, the PS III is also superior to the Hapi in these respects, but the PS Linear Control Center is not as good as the high-level section of the Hapi, so that the straight-through sound of the latter is somewhat superior to that of the PS system. What all this adds up to is that the Hapi Two is the best all-in-one preamp/control box we've tested that you can buy right this minute, as we go to press—regardless of price.

Considering that we first told you about the Hegeman preamp more than a year and a half ago and that the circuit has gone through only very minor modifications since (and none lately), it has held its high ranking with remarkable consistency on rapidly shifting ground.

Precision Fidelity C7

Precision Fidelity, 1238 Green Street, San Francisco, CA 94109. C7 cascode preamplifier, $499.95. One-year warranty (tubes 90 days). Tested #79840, on loan from manufacturer.

The first thing that needs to be pointed out about this outstanding and attractively priced audio component is that it isn't really a complete preamp/control unit. It has no high-level stage. It consists basically of the excellent phono stage of the C4, with the gain boosted to 50 dB at 1 kHz, followed by a level potentiometer (one per channel, no double-deck stereo volume control, no conventional balance control) and facilities to select various high-level sources (tuner, aux, tape monitor) without additional amplification. If these sources don't have enough output to drive your power amplifier—forget it. In most cases it works out just fine, as long as you remember that those potentiometers create a fairly high-impedance source, as high as 7000 ohms depending on their setting, so that you need low-capacitance output cables to avoid rolling off the highs.

The phono sound of the C7 is slightly different and, if anything, even more appealing than that of the C4, which did very well indeed in our previous series of tests. The C7 has a totally smooth, unstrained and edgeless sonic character, with tremendous dynamic headroom, giving the impression of a sweet, mellifluous, luxuriously cushioned quality that some listeners find more "musical" than the more transparent, analytical and airily detailed sound of the best transistor preamps (Cotter, Hegeman, PS III). The C7 is very easy to listen to, perhaps just because it wraps this pleasant softness around the music—almost a kind of veil—but for that very reason it must ultimately be judged less accurate than the solid-state contenders for top ranking.

At first we suspected that some kind of RIAA equalization error was responsible for this "personality" of the C7, but that didn't turn out to be the case. We measured virtual perfection from 1 kHz on up; the lower frequencies were uniformly depressed along a -0.3 dB shelf, dipping to -0.5 dB at 150 Hz—certainly nothing dramatic. Another possibility was high-frequency mixing or cross talk between the stereo channels, which tends to create a softening effect, but the measured separation was excellent. Square waves also looked quite good at all frequencies; in fact, we detected no glaring anomalies of any kind, anywhere. Correlating the characteristic sound of the unit with measurable data would obviously require extensive detective work, with no guarantee of success.

None of the above should be interpreted as a lack of enthusiasm for the C7. If those three solid-state preamps didn't exist, the C7 would be in our "Reference A" system, and we would be happily listening to it without much awareness of what we were missing. It's that good.

PS III and PS LCC

PS Audio, 3130 Skyway Drive, #301, Santa Maria, CA 93454. PS III Phono Preamplifier, $222; PS Linear Control Center (LCC), $240; optional rack panel, $25. Five-year warranty. Tested #0393 and #0362L, on loan from manufacturer.

Paul and Stan (the P and S of PS Audio) have come a long way since the PS II plug-in phono stage we reviewed two issues ago. The similarly conceived and configured but vastly improved PS III is now just a very small step behind the Cotter PSC-2 in sonic accuracy (at a ridiculous fraction of the cost!), and the Linear Control Center, though not as amazingly transparent and neutral, also sounds truly excellent. The entire system, which will set you back all of $487 if you buy the single-piece rack panel that ties together the two little self-contained boxes, is our obvious and unequivocal selection for the front end of "Reference B." It could even qualify as a temporary "Reference A" preamp while waiting for the entire Cotter System 2 to become available on a regular production basis.

The only difference in sound between the Cotter and PS phono stages is a very slight tendency by the latter to displace high-frequency information forward, toward the listener. In more exaggerated form, this anomaly ends up as overbrightness or hardness; in the case of the PS III it's barely perceptible. The two units have virtually the same sound, which is in effect no sound; it takes electrostatic speakers and a very fine power amplifier to zero in on the differences. It's interesting to note that both operate in the current mode and both are passively equalized for the RIAA playback characteristic. Talk about a convergence toward functional realities . . .

As long as we're talking about RIAA equalization, it should be pointed out that the PS III is right on the button, ±0.0 dB, from about 50 Hz on top to ultrasonics; at 20 Hz the error is a staggering -0.2 dB. (That includes our possible measurement error.) So you see, all you engineering geniuses, it can be done for no money at all, with just a little bit of devotion. All other things check out equally well on the PS III, including preequalized square waves, cross talk, you name it. All in all, we're unaware of a third phono preamp that isn't some distance behind the PS III, at any price. That statement applies, however, only to the "moving magnet" position of the gain switch; the "moving coil" position is no substitute for a good transformer or even a good cheap pre-preamp like the Marcof. We almost wish, for the sake of this breakthrough product's credibility, that the extra little MC feature hadn't been thrown in. It sounds bad, whereas at normal gain the PS III sounds incredible.

The LCC unit has a few minor problems, including a bit of cross talk, all tending to create a very slight veiling effect. Even so, the PS III *cum* LCC still sounds cleaner and more transparent than other complete preamp signal paths we've tested, with the exception of the Cotter and the Hegeman. For the money, that's still incredible. Of course, if your power amp has level controls (only a few do), you can eliminate the LCC altogether. The PS III has plenty of gain.

The construction of the PS units is a mixed bag of credits and demerits. The metalwork has only very recently been changed to anodized aluminum; the earlier painted boxes were simply atrocious in appearance, with misaligned screw holes and barely fitting tops and bottoms. Inside one discerns an obvious attempt to spend the money where it does the most good and skimp on whatever is less important. It's an intelligent trade-off but some purists who won't be able to resist the sound will miss that reassuring mil-spec look. The basic question is whether the sound will be exactly the same after years of daily use. That remains to be seen.

For the moment, we're all admiration with hardly any reservations. This kind of money in the past has always bought a formatted sound with characteristic colorations you could either like or dislike; now for the first time, achromatic preamplification is cheap.

Recommendations

This time, they're obvious and require no further explanation if you've read the reviews above and in the previous issue.

Best preamplifier so far, regardless of price: Cotter System 2.

Best preamplifier per dollar: PS III with PS LCC.

Reference System Revisions and Updates

Reference A is the best we've been able to put together so far, regardless of price. Reference B is the best we know of per dollar. Both systems have undergone some changes since the last issue.

The rationale behind each of these two very different reference systems was presented and analyzed at some length in the last two issues. By now it should suffice to list the various components, all of which have been reviewed either in previous issues or in this one, and to indicate which are new selections.

Reference A

The caveat that prefaced our selections in the last issue no longer applies. This is now a system that anyone can assemble and get the most out of, without test instruments. All it takes is a check book.

Speaker System

From 100 Hz on up, a new electrostatic unit: Sound-Lab R-1 ($2200 the pair).

Below 100 Hz, the trusty old Janis W-1 ($1350 the pair).

Power Amps and Crossovers

To drive the Sound-Lab, a choice of two new amplifiers: Bedini Model 25/25 ($650) for the ultimate transparency but limited headroom; JVC M-7050 ($1500) for the best combination of transparency and power.

The Janis woofers are driven, as before, by a pair of Janis Interphase 1 bass amplifiers with built-in 100 Hz electronic crossovers ($495 each, $990 the pair).

Preamplifiers, Interface and MC Step-Up

No change here; the Cotter System 2 is still our choice. The chain is: MK-2L transformer ($600) into PSC-2 phono stage ($500) into CU-2 control unit ($1450, if and when available) into NFB-2 noise filter/buffer ($450) into the crossover input. The last three are powered by the PW-2 power supply ($280).

Phono Cartridge

The Koetsu moving-coil pickup (now down to $750) is still our recommendation to subscribers; we're using the experimental hybrid discussed on pages 46-47 of this issue.

Tone Arm

Nothing has come up so far to replace the Fidelity Research FR-66s twelve-inch arm ($1250).

Turntable

Again no change; the Cotter B-1 system remains our choice. Currently it comes only with the Denon DP-80, factory-installed (approx. $2500 to $2600, depending on dealer); we're still using the almost unobtainable Technics SP-10 Mk II in our B-1.

* * *

This latest revision of Reference A now costs between $13,000 and $14,000 at retail, a significant drop from the previous version. The saving is a reflection of considerable simplification without any compromise in sonic performance.

Reference B

We almost envy those who are shopping in this category at the present time; never before has this kind of superior audio equipment been available for this kind of money—adjusting, of course, for inflation. If you're starting from scratch, consider yourself lucky.

Speaker System

If accuracy and lack of colorations are top priority: Vandersteen Model IIA ($940 the pair), not to be confused with the Model II previously recommended.

If sheer SPL capability and dynamic range are very important: DCM Time Window ($660 the pair).

Power Amplifier

The new Amber Series 70 ($459.95) replaces our previous selection.

Preamplifier

Another change: PS III Phono Preamplifier ($222) with PS Linear Control Center ($240) and optional rack panel ($25).

MC Step-Up

The same as before, but in the new and improved version: Marcof PPA-1 ($124.95).

Phono Cartridge

No change: Fidelity Research FR-1 Mk 3F ($230).

(Our previous alternate recommendation of the Win Laboratories phono transducer system, which eliminates the need for a preamp and a step-up device, must be held in abeyance until we find out what the latest production version is all about.)

Turntable with Tone Arm

Entirely new recommendation: Kenwood KD-650 ($400).

Substituting the armless Kenwood KD-600 ($350) and mounting the JVC UA-7045 arm ($250) on it would be even better, though possibly beyond Reference B budgeting.

In either case, the use of the Cotter B-2 isolation platform ($195) may be necessary for best results in certain installations.

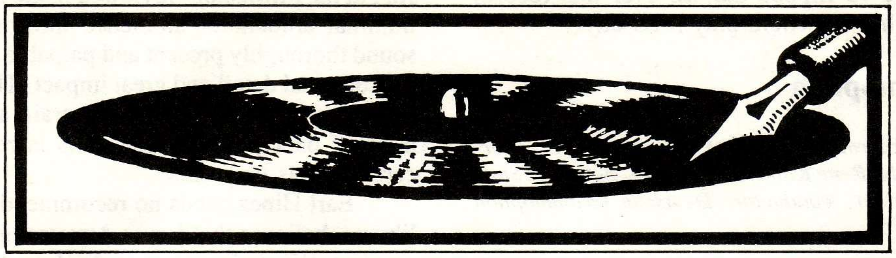
* * *

Depending on the alternate choices opted for, this latest version of Reference B costs between $2300 and $3000 at retail, which isn't half bad considering the greatly improved sonic performance and the inevitable rise in prices since last time.

Page-Counting Subscribers, Please Note:

This would be a 76-page issue if all type sizes were exactly the same as last time. We decided to set all equipment reviews and certain articles in smaller but still highly readable type, in order to save pages and combat our insanely high printing and mailing costs.

Records&Recording



Editor's Note: *Max Wilcox, who has had this column as his bailiwick since our earliest issues, has been too busy recording, both in the United States and abroad, to have time left for writing us a new article. Don't worry, though; he'll be back. Meanwhile we continue to talk about the records we like to pull out when we want to check out a new system or new components.*

A Discography for the Audio Purist: Part III

As you know from the two previous installments of this series and our original explanation of the ground rules, we don't publish "record reviews" here in the conventional sense. That would mean, inevitably, a mixture of good and bad, approval and disapproval. We hope to introduce such reviews in future issues as **The Audio Critic** expands, but this discography includes only records we have found exceptionally interesting sonically, either for utter naturalness or for other characteristics useful in evaluating audio equipment. We run across some very fine records, musically and/or audiowise, that we have no specific reason to mention here. So there's absolutely no stigma attached to noninclusion of a particular label or disc. This is a very short list.

That said, we feel we must still come back once more to the almost universally adulated new digital recordings and restate, in anticipation of virtually certain protests, why we aren't ready yet to join the worshipers. As we explained, the present sampling rate of only 50,000 samples per second results in a somewhat degraded, electronic-sounding top end. In other words, at the present state of the art, we believe we can hear the digitizing process. The new brass quintet and trumpet recordings on the **Delos** label have given us our most recent proof of that belief. These were recorded with strictly purist techniques, using a pair of B & K microphones, very much like the Mark Levinson brass album we reviewed four issues ago (Vol. 1, No. 4). The main difference was the Soundstream digital system used by Delos as against 30-IPS analog recording used by Mark Levinson. Even though the Delos records are superior in many ways—signal-to-noise ratio, spatial presentation, general musicality of production, and the quality of brass playing—the Mark Levinson record yields the more natural and believable brass sound, at least to our ears.

At the same time, there's no doubt in our mind that with a sufficiently high sampling rate and a sufficient number of bits, digital recording can far surpass all analog systems. The day will come. At this point, however, the few records we're adding to our discography are all analog.

Desmar

Schubert: Sonata in A Major, Opus Posthumous. Richard Goode, piano. Desmar SR-6001 (made in 1978).

Just because Max Wilcox is our Contributing Editor we still can't, out of sheer journalistic impartiality, ignore his recording work when it's this good. Yes, this is the sound of the "new" Max Wilcox—two Schoeps omnis, 30 IPS, minimal console electronics. The piano sounds totally natural, rounded and beautiful, not too close and not too far away. Max doesn't like that steely ping of the upper strings that you get with very close-miked piano recording; personally we would prefer just a touch more of it, but that's quibbling. The piano and the space are all there, completely audible in every detail, and that's what counts.

This is one of the great works of Schubert's "late" period, if the word is at all applicable to a composer who died at 31. (We firmly believe that had penicillin existed in the early nineteenth century, Schubert would have lived to surpass Mozart and Beethoven in stature. He was getting there.) The music speaks in long, drawn-out sentences and paragraphs, but with incredible melodic and harmonic invention. You don't want any of the movements to end; it's all too beautiful for interruption, even with something just as beautiful.

Richard Goode is a marvelous musician to whom this expansive idiom is as natural as speech. He maintains the long line without a moment of sagging, all the time delighting you with his exquisitely simple phrasing, superb voice leading, lovely tone, and unerring rightness of expression. A few lapses of taste, a bit of egocentric virtuosity, an occasional italicization of the wrong detail could make a shambles of this leisurely and subtle work. Richard Goode plays it like great chamber music, first things first, the forest before the trees.

If you think we've flipped our lid over this record, you're absolutely right. We could play it all day.

Deutsche Grammophon

Johann Strauss: Die Fledermaus (complete operetta, 4 sides). Hermann Prey, Julia Varady, Rene Kollo, Lucia Popp, Bavarian State Orchestra, Carlos Kleiber, conductor. Deutsche Grammophon 2707 088 (made in 1976).

This is a big multimike production, not without console shenanigans, but the very best of the genre, with a tremendously real stage that has genuine sonic breadth and depth. Everything is three-dimensional. You can close your eyes and follow the singers around. What's more, the dynamic range is excellent and there's never any strain, not even on soprano high C's. A very stylish recording job, everything considered; it remains to be seen whether *anyone* can do this sort of thing better with fewer channels and fewer microphones.

Many people don't realize just how great this music is. Melodically, of course, it's the apotheosis of Viennese three-quarter time, but that alone would make it merely delightful, not great. The ensemble writing, however, has an almost Mozartian perfection and the orchestration is superb. There just isn't any better *light* music than this.

The performance could be described as near great, lacking only the ultimate degree of rip-roaring, uninhibited exuberance, such as we remember from some classic predecessors. Instead, it's lilting, elegantly effervescent, thoroughly idiomatic, very precisely controlled. Carlos Kleiber conducts it as if it were great music, not just fluff, and the singing is mostly wonderful. One of our favorites.

M & K RealTime

The Magnificent Basso (assorted works by Carl Loewe, Mozart and Verdi). Michael Li-Paz, basso, with Zoltan Rozsnyai, piano. M & K RealTime RT-102 (made in 1978).

Direct to disc and miked without any room sound whatsoever, this rather perverse production is useful to the audio equipment reviewer because of its very perversity. This kind of recording is totally chameleon-like; the sound of the basso's voice changes in direct proportion to the colorations inherent in the equipment under test, and his apparent position is entirely dependent on the radiation characteristics of the speakers. This would not be as clear-cut were the voice more spaciously and luscious-beautifully recorded. It also helps that the disc is superbly quiet and has great dynamic range. Quite a tool.

It's almost an irrelevance to state after this that Michael Li-Paz is a good basso who understands the music he sings. For musical enjoyment, he should have been recorded by Deutsche Grammophon.

* * *

''Fatha''—Earl ''Fatha'' Hines Plays Hits He Missed. Earl Hines, piano; Red Callender, bass; Bill Douglass, drums. M & K RealTime RT-105 (made in 1978).

Much more natural than the above, although lacking the uncannily lifelike spatial characteristics of the Proprius *Jazz at the Pawnshop* (reviewed in the last issue). Despite the minimal amount of ambience information, the musicians sound thoroughly present and palpable, with superbly etched instrumental detail and great impact. The sound is absolutely clean, top to bottom, with no strain at any level and not a trace of background noise. This is realism. Direct-to-disc certainly has its points.

Earl Hines needs no recommendation to anyone who, like us, believes that Louis Armstrong was and remains the greatest. ''Fatha'' goes right back to the early days of Louis and all those old-time good 'uns, the very roots of jazz. Here he plays more modern stuff, but the style is still entirely his own and it never fails to swing. Highly recommended.

Mobile Fidelity Sound Lab

This company borrows existing original master tapes, mostly of big hits in the rock-pop idiom, and manufactures new discs from them with immense care: new lacquer masters cut at half speed, the best vinyl money can buy, super pressings, highly protective packaging, etc. That means they have a presold market, since every Fleetwood Mac enthusiast with an expensive stereo system is a potential customer; from our point of view, however, most of this music is unrelated to high fidelity, having been made on a zillion tracks with every electronic production trick in the book, more processed than Kraft cheese. There are a few exceptions, however. What's more, the Mobile Fidelity treatment has revitalized a number of worthwhile pop classics. For example, the Beatles' *Abbey Road*, their last and undoubtedly one of their two or three best albums, sounds almost as if it had been recorded with present-day techniques and is in every way a new experience on the latest Mobile Fidelity release, received just before we went to press. Still, it isn't an audio tester's kind of record. The one below is.

* * *

Pink Floyd: The Dark Side of the Moon (originally recorded in 1972 and 1973). Mobile Fidelity Sound Labs MFSL 1-017 (made in 1979).

This record has always been our favorite exception to the rule that you can't judge audio equipment on rock music. The multimike/multitrack/multiprocessor technique is applied with tremendous sophistication by these people, with an end result that may be far removed from real-world sounds but is just as cleanly etched and impactful, free from anything that resembles ordinary distortion, in addition to being considerably larger than life. The opening ''heartbeat'' passage has become the standard bass test of the audio salons. On top of it, this is very listenable, musical rock despite the deliberate touches of weirdness—at least to our ears, which we must admit go into their protective shutdown mode after the first two bars of typical heavy metal.

We're able to hear details and subtleties in the Mobile Fidelity version that we weren't aware of even on the original EMI pressing, let alone the USA copies. For the first time, we can clearly make out every dirty word as well as all sorts of instrumental textures. Great fun and proof positive that the ''Original Master Recordings'' concept of this company isn't just a hype.

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The Audio Critic

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We take our most critical look yet at phono cartridge design and performance.

We hope to report the final solution to our long-standing problem of standardizing a meaningful test for resonance and acoustical breakthrough in turntables and tone arms.

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