Volume 2, Number 1 Winter/Spring 1979

The Audio Critic®

Retail price: $6

In this issue:

We publish Part I of the transcript of our all-day seminar on the State of the Art, where the juxtaposition of some of the best minds in the audio world resulted in the ultimate hi-fi bull session.

Our medium-priced, best-sound-per-dollar Reference B system is totally revised in response to new component developments.

An important guest article lifts the veil of obfuscation from stylus design and tip geometry.

Plus, of course, our usual quota of equipment reviews (heavy on speakers and preamps), along with our regular columns.

The Audio Critic

Volume 2, Number 1 Winter/Spring 1979

Editor and Publisher	Peter Aczel
Contributing Editor	Max Wilcox
Graphic Designer	Dick Calderhead
Business Manager	Bodil Aczel
Assistant to the Publisher	Elizabeth Tinsley

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

The Audio Critic® is an advisory service and technical review for consumers of sophisticated audio equipment. It is published periodically, at intervals averaging approximately four 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.

Contents

Publisher's Note

Here we go again. We're late. Not as late as we were with Volume 1, Number 6, but late enough to get some of our subscribers upset all over again. We're well aware of it. In fact, we have a feeling as we bring you this unprecedentedly fat 80-page issue that these subscribers might have been happier if we had split it into two 40-page issues published twice as fast—in which case they would have been paying twice as much for the same amount of information but would have blissfully cycled through two surges of mailbox anticipation and gratification instead of one. And, of course, we would have been fulfilling two issues' worth of subscription fees with just about the same effort and investment. Pretty stupid of us not to give people what they want, especially when it would save us work and money. What do you think?

Well, the way it happens is a vicious circle. It gets to be a little late; meanwhile more and more interesting new information piles up; then we just can't stand not publishing all we've found out, since the following issue is too far away. So, when we finally go to press, the new issue reflects our most up-to-date point of view, but our publishing schedule is shot to hell and we're giving away much too much for one sixth of a subscription. We now realize that we can't go on like this issue after issue and that we must sooner or later restructure our entire format and subscription package. One possible solution would be to separate the theoretical material from the equipment ratings, publishing the latter with much greater frequency, and the theory in larger chunks and greater depth, but separately and less often. We're in no position, however, to make any such conversion in the near future; it will take place, if at all, sometime in 1980. You'll be notified in plenty of time to think about it, and the conversion formula will be such that you'll end up with more information than you originally paid for, but we won't end up being the bad guys who deliver too much too late.

Until then we want to reiterate our belief that, whether **The Audio Critic** *comes out bimonthly, quarterly or (God forbid) semiannually, anyone who reads and assimilates every word in every issue possesses a devastating superiority—both in real-world, no-bull audio knowledge and in protective consumer reflexes—to anyone who reads only the hi-fi slicks and/or the undergrounds with equal diligence. If we didn't believe that, we wouldn't even bother to publish the next issue.*

* * *

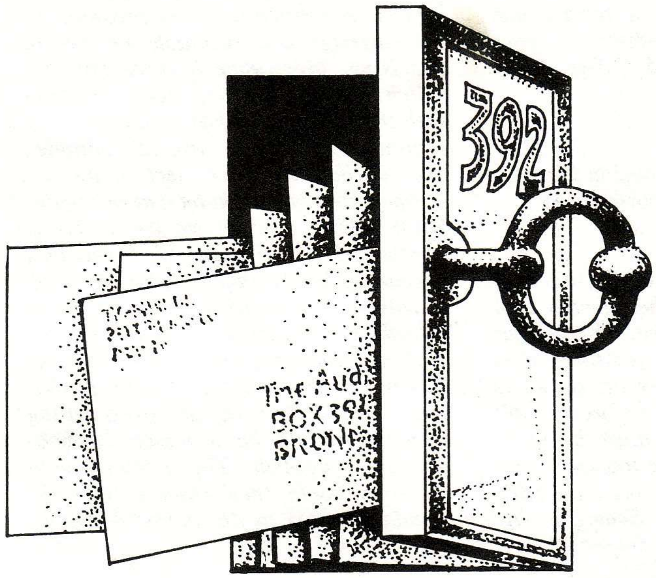
You should be aware that individual copies of **The Audio Critic** *are now being sold over the counter in a small number of selected audio stores. We feel this will give us needed exposure and an expansion of our marketing base; at the same time our first obligation is still to our subscribers. Therefore the over-the-counter price has been set 20% higher than the subscription price, and subscriber copies will always be mailed first when a new issue is published. Subscribers will also have special advantages and privileges if and when we convert to a new format as mentioned above.*

* * *

Again we must remind you that where-the-hell-is-my-latest-issue letters, if no such issue is off the press yet, will be answered only when accompanied by a stamped and self-addressed envelope. We don't have fifteen cents to spend on explaining to someone that an unpublished issue is very difficult to mail.

Box392

Letters to the Editor



To make room in this issue for the voluminous transcript of our State of the Art seminar, we're cutting back the space allotted to a number of our features, including this column. (How much small print can we run, after all?) That means we're publishing only those letters this time that demand immediate attention in our judgment, saving others of "evergreen" subject matter for future use. As a matter of policy, any letter of general interest or specific concern to our subscribers (rather than just to the letter writer) stands a good chance of being printed here; letters 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.

First, the letters on what currently appears to be the premier subject of our editorial correspondence, phono tracking alignment.

The Audio Critic:

Being avid readers of your esteemed publication, we would like your readers to be told of a European pioneer in the field of lateral tracking geometry. The mathematical proof of the distortion effects of lateral tracking error is 40 years old rather than 37 years. Erik Lofgren published his first paper on this subject as early as 1929 in the Swedish journal *Radio*.

We enclose a copy of Professor Erik Lofgren's German-language paper in the then-young *Akustische Zeitschrift*, Nov. 1938—three years prior to Mr. Baerwald's paper. We have attempted an English summary which we hope you will publish.

Many thanks for Part III with the excellent alignment instructions! Do keep on spreading the light!

Yours sincerely,
Sven Eriksson
Lars Backlund
Ingenjorsfirma Sven Eriksson
Johanneshov, Sweden

Yes, indeed; had no other paper been published on the subject of lateral tracking error after Lofgren's, tone arm designers would still not be without a mathematical model for specifying effective arm length, offset angle and overhang correctly, although the specified values would differ by trivial amounts from those dictated by the Baerwald criteria. Therefore, in that sense, it's true that Lofgren beat Baerwald to the punch by three years.

It must be pointed out, however, that Baerwald was totally conversant with the Lofgren paper and gave it full credit, while arguing that his definition of minimum distortion over the recorded area of the disc was preferable to Lofgren's and that his use of the Chebyshev approximation to calculate correct tone arm design parameters was somewhat superior to the method of least squares suggested by Lofgren. Furthermore, the Baerwald paper is by far the more comprehensive and profound of the two; for one thing, it spells out more clearly and emphatically the crux of the matter: that tracking error generates FM distortion, which is more annoying than ordinary harmonic distortion.

Since we never published a summary of Baerwald, we don't see a compelling reason for publishing a summary of Lofgren, either, but we certainly appreciate having the full text of the original 1938 paper in German through the courtesy of our Swedish friends. Whether the original insights came from Stockholm or Cleveland, Tokyo has certainly had enough time by now to heed them. Thank you, Messrs. Eriksson and Backlund, for understanding and caring.

—Ed.

The Audio Critic:

I would be very interested to know what the tracking error is at the extreme outer groove corresponding to your Optimum Overhang and Offset Angle Chart? . . .

Sincerely,
G. Bearman
Chairman & Managing Director
Mayware Ltd. (Formula 4)
England

The absolute value of the tracking error would depend on the effective arm length: the longer the arm, the smaller the tracking error. The question, however, is essentially irrelevant; as we've pointed out many times before, it isn't the tracking error that must be minimized but the ratio of the tracking error to the radius at which is occurs. That's what Lofgren, Baerwald, Seagrave, etc. are all about. Thus the tracking error is inevitably largest at the outermost groove, and that's the way it has to be.

If this query has anything to do with the Formula 4 Mk III tone arm, we must add that its offset angle is slightly incorrect and its specified overhang more than slightly so. An arm with an effective length of 229 mm should have an offset angle of 24° 5' (not 23° 40') and should be mounted with an overhang of 18.1 mm (not 20 mm) if it is indeed optimized for a 12-inch LP disc as the Formula 4 is claimed to be. Furthermore, the inner zero-tracking-error point should be at a radius of 66.04 mm (instead of 63.50 mm). Not as a matter of opinion but in accordance with natural law. (Our apologies if we misinterpreted the intent of the question.)

—Ed.

The Audio Critic:

Enclosed is some material by Percy Wilson on VTA and elliptical versus spherical styli that may interest you, if you have not seen it already.

In the June 1964 *Gramophone* Wilson puts his finger on a difficulty with any attempt to adjust the VTA: tilting the cartridge or arm to correct the VTA also tilts the stylus away from perpendicular, so that the stylus no longer fits the record groove. The line contact of the new stylus shapes would thereby be upset. Until the VTA is standardized there would appear to be no fully satisfactory solution . . .

Sincerely yours,
F. Brock Fuller
California Institute of Technology
Pasadena, CA

Your letter arrived very shortly after we had started to worry about the same thing ourselves. You're undeniably right, and our current thoughts on the subject are written up in the preamble to the cartridge and turntable reviews in this issue. Meanwhile it turns out that Mitch Cotter has also been wrestling with the problem; he has outlined to us a solution so radically original, complex and daring that we feel quite incompetent to comment on it at this point. We'll believe it when we see it. To us simple folks, a stringent VTA standard appears to be the shortest possible route to satisfaction, at least as far as future records are concerned. Since the number of decision makers on the record cutting side of the phono industry isn't all that large, we can't see why at least a cutting standard should be a major "technopolitical" problem. The pickup manufacturers could then comply with the standard or not, as the market demands.

—Ed.

The Audio Critic:

In your Spring through Fall 1978 issue you published my letter to Mr. Bill Carter of Australia, which he submitted to obtain your response to my views on lateral tracking error in pickups. You kindly printed my whole argument, but proceeded not-so-kindly to dismiss it as 'casually condescending speculation' which ignored the investigations of Baerwald and others. Now I am aware of the papers to which you refer, and of the fact that some aspects of tracing distortion may be regarded as equivalent to a form of frequency modulation. (Doppler distortion in loudspeakers is another example of this interchangeable viewpoint, where time-displacement sidebands closely resemble ordinary IM products.)

The reference in my letter to time errors between the two groove walls was not made in ignorance of the Cooper/Woodward theses, but in simple response to your own argument on the matter, spanning pages 52/53 of your July-September 1977 issue. Perhaps it was not your intention, but the wording there seems to imply that the 'time smear' of 12.5 uS with which you were concerned involved the strictly stereo aspect of the signal. You stated that it 'can blur the focus of a stereo signal to some extent. (This is not an unreasonable figure to assume.)'

My reaction was to this simple point. I argued to Mr. Carter—and stick to my guns—that the suggestion as put *is* unreasonable. A phase error between channels corresponding to such a minute time difference would *not* blur the focus of a stereo signal. I readily accept that I misconstrued your comments, and that my rewording of your explanation must have seemed puzzling, but I humbly suggest that the initial fault was in your own misleading use of words. I know from bitter editorial experience how easy it is to say or imply the wrong thing, and I see that even in the disputed letter I referred at one point to the 'two grooves' instead of the 'two groove walls'. We must all try to be more careful.

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

Agreed. We must all try to be more careful. We'd be the last to claim that our writing is so simple and lucid that further editorial effort couldn't make it simpler and more lucid yet. But as we depart uttering these humilities, we're strongly tempted to turn around in the doorway, like Peter Falk doing one of those delayed exits in the Columbo series, diffidently raise our right hand, and say, "Just one more thing . . ."

Just one more thing, Mr. Crabbe. Didn't you write in your letter to Bill Carter that the audibility of tracking-angle error is in your opinion due to "old-fashioned harmonic or IM distortion"? Old-fashioned, right? Would a person whose perception of the subject is informed by the Baerwald, Bauer, Woodward and Cooper analyses use that word? Are time-dispersive automodulations of the signal the same as good old THD and IM? Oh, I see. Just another little carelessness in the use of words. Well, thank you very much, sir. I'd better be going now . . .

We're inclined to believe that Columbo would arrest Mr. Crabbe the very next morning for intellectual weaseling in the first degree. But until a fair trial, the Anglo-Saxon presumption of innocence must apply.

—Ed.

* * *

And now, just a few more letters in the miscellaneous category.

The Audio Critic:

I have noticed that among all the amplifiers and preamplifiers that you have tested, you have never mentioned McIntosh, even though this is one of the most widely known and sold brands. Did you have a particular reason for this? Is the equipment so bad that it does not even warrant a mention? Or are audiophiles just biased against Mac?

Sincerly yours,
Lester F. Keene
Cocoa, FL

McIntosh established its original reputation with outstanding vacuum-tube equipment back in the late forties and early fifties, plus superb dealer relations over the years and the merchandising attitude that the customer is always right. Such a solid foundation is virtually unshakable in the marketplace, even after years and years of engineering mediocrity. We don't know of a single SOTA-oriented audio professional, however, who believes that McIntosh is in the forefront of the purist/perfectionist sector today—or was even a decade ago. Unfortunately, to prove this with our own laboratory and listening tests, we'd have to purchase the equipment, since McIntosh doesn't believe in lending stuff to noncommercial reviewers. And if we bought it, we wouldn't know where to sell it after the tests. In the audio circles where we move, nobody wants it or even knows anyone who might.

—Ed.

The Audio Critic:

Thank you for reviewing our 10/24 subwoofer. The following may be of value to interested readers:

(1) We have no dealers. For information please write The Bass Mint, Box 153, Powell, OH 43065. SASE's are very much appreciated.

(2) The price of the 10/24 is $250 apiece, $475/pair, *plus shipping.* Shipping charges are not refundable under the terms of our 30-day trial period.

(3) We will not make specific brand recommendations on crossovers, power amps, or speakers to go with the 10/24. That is the province of *The Audio Critic* and other magazines. We do recommend low-level "electronic" crossovers and separate bass amplification (as opposed to high-level passive devices), subsonic filters, and good turntable isolation.

(4) We have no phone listing at this time. Due to our limited manpower we just can't afford to be on the phone all the time. Callers trying to reach me at home will encounter my answering device. Please write.

Thank you.

Sincerely,
Ed Cottle
President and Stock Boy
The Bass Mint
Powell, OH

Maybe that's why your subwoofer is correctly aligned and so many others aren't. While you're doing your engineering homework, those other designers are busy gabbing on the phone.

—Ed.

The Audio Critic:

I am pleased with your evaluation of the sonic virtues of the H-3aa power amplifier but do take exception to two of your assertions.

1. The power tubes I use (6LF6) are being manufactured in the USA by GE and Sylvania. They are also being made in Japan and Yugoslavia. I have been informed that they will be around for many years. In your Vol. 1, No. 3 issue, page 4, you wrote that the Berning hybrid tube amplifier using 6LF6 tubes will be manufactured by Audionics. Also please note: The Acoustat X ($2200) uses 6HB5 tubes, which is also a TV tube. This tube was used in my H-3 stereo amplifier in the 1960's. The Beveridge 2SW-1 ($7000) uses 40KD6 tubes, also a TV tube. My point is that these two types are of even older vintage than the 6LF6 tube, yet you did not caution a buyer of the Beveridge system about their not being able to obtain them in a few years. I am sure if tubes do a better job in an amp or preamp they will always be available from one source or another.

Before taking up the second subject that makes me unhappy I would like to digress for a minute, if I may. Before transistor amplifiers the rated impedance of most high fidelity speakers was 16 ohms. In Britain practically all speakers were 15 ohms. There were sound reasons for this as, all other things being equal, a higher impedance speaker is more efficient and the crossover design is not as complex. We are referring, of course, to moving-coil speakers.

Electrostatic speakers are inherently of high impedance and this is lowered by means of a transformer. The Acoustat and Beveridge speakers use a different approach. They do not use a transformer; instead they employ very high-voltage amplifiers to drive the speakers directly. The KLH-9, Quad, and Koss are examples of speakers using transformers. In general, the lower the turns ratio of the transformer the better the speaker because of tighter coupling and other factors that I will not go into here. The KLH-9 impedance is 16 ohms, the Quad 15 ohms and the Koss 4 ohms. The KLH-9 and Quad were designed for tube amplifiers, the Koss for solid state.

The reason for the lower impedance of speakers today is, of course, the fact that transistor amplifiers, being voltage limited, provide more power for such speakers. As an interesting aside you implied in your review of the Tangent RS2 (Vol. 1, No. 5, page 25) that it was an inefficient speaker as you were able to make the Levinson ML-2 clip on it with a master tape of piano music. On the other hand, I can make the Tangent RS2 play very loud with the H-3aa. The reason for this is simple: The impedance of the RS2 at 70 Hz, for example, is 11.5 ohms; at 500 to 2000 Hz it is 9 ohms, and it rises steadily to over 20 ohms at 6 kHz, which is well above the fundamental tones of the piano. With the ML-2's 14 volt maximum voltage rating you can see that there is very little power to drive the Tangent. End of digression.

2. Many audiophiles are using the H-3a and H-3aa with electrostatic speakers such as the KLH-9's that keep their impedance high up into the upper range and also with the Quad, which does fall to low values but nonetheless sounds extremely good. For owners of double Quads I recommend wiring them in series and, if I may be allowed to boast a little, they do sound fantastic.

Thank you for allowing me to comment.

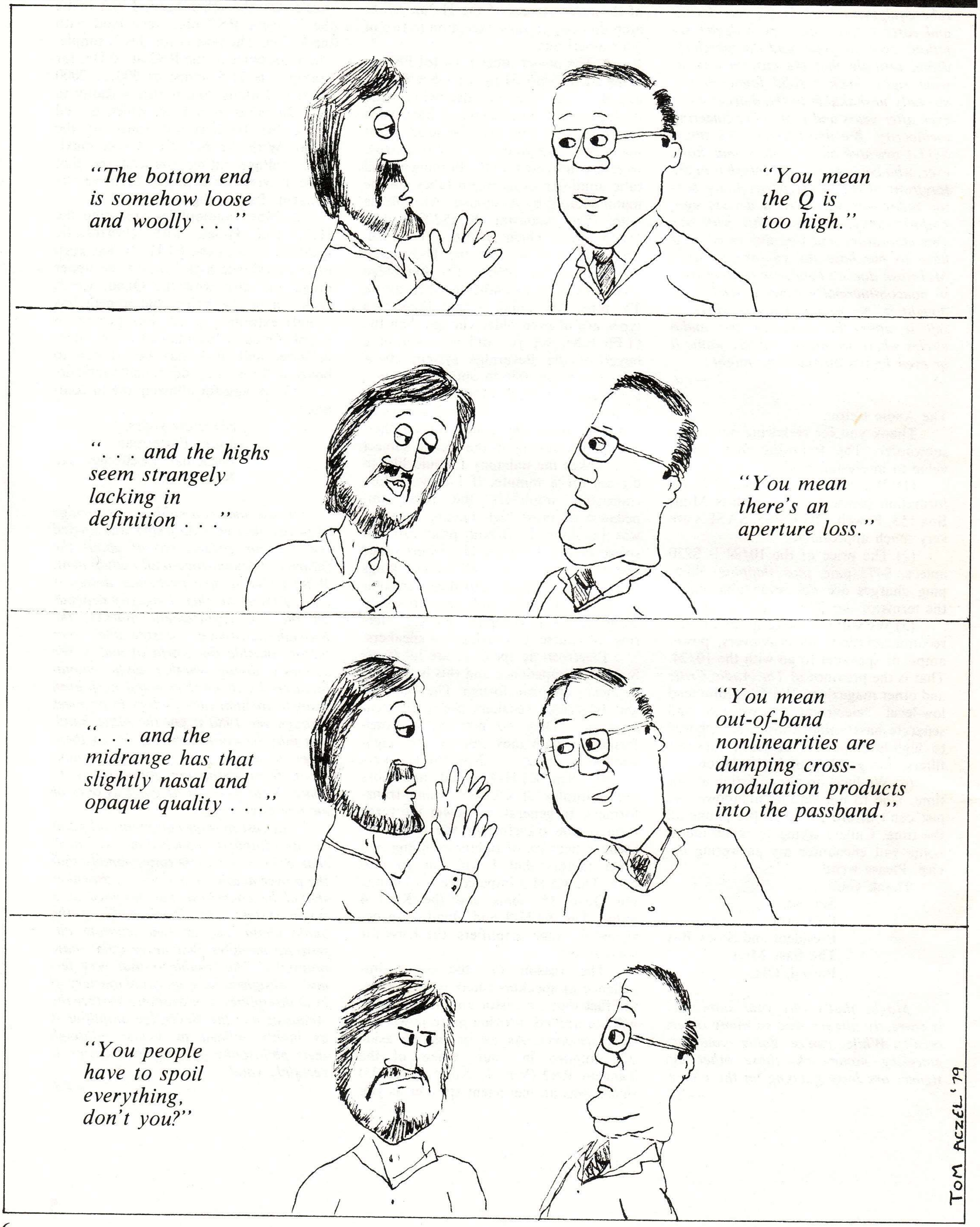
Sincerely yours,
Julius Futterman
Futterman Electronics Lab
New York, NY

It was unquestionably a miscarriage of justice that your amplifier was singled out for our general caveat about the future of vacuum-tube audio equipment. What's true of one particular design is true of them all: their longevity depends on the TV replacement market, the Russian aerospace industry and other factors outside the world of audio. We seriously doubt whether audio manufacturers by themselves could keep even a single vacuum tube factory in business through the 1980's. On the other hand, you may be quite right insofar as these other demands may preserve vacuum tubes from extinction for decades to come. Your guess is as good as ours or anyone else's.

The rest of your comments all point to an implicit conclusion we have shared for quite some time, namely that the power amplifier and the loudspeaker should be conceived and designed as a single system, the "back end" of the audio chain, not as two separate all-purpose modules that never quite mate optimally. The trouble is that very few audio designers have an equal mastery of both disciplines. For example, neither the Acoustat nor the Beveridge amplifier is as highly refined as yours, although their philosophy of integrated design is certainly valid.

—Ed.

In Your Ear


"The bottom end is somehow loose and woolly . . ."
"You mean the Q is too high."
". . . and the highs seem strangely lacking in definition . . ."
"You mean there's an aperture loss."
". . . and the midrange has that slightly nasal and opaque quality . . ."
"You mean out-of-band nonlinearities are dumping cross-modulation products into the passband."
"You people have to spoil everything, don't you?"
TOM ACZEL '79

A Challenge to All Critics of *The Audio Critic*

By Peter Aczel
Editor and Publisher

Any audio journalist, equipment designer, manufacturer or other audio practitioner who delights in bad-mouthing us, or maybe just automatically contradicts us at every turn, is hereby given the chance to destroy our credibility in front of all our readers. Or forever hold his peace.

We continue, for indexing purposes, to number our editorial topics sequentially, picking up where we left off in the previous issue; this is not to be construed as any suggestion of a serial story. What follows is complete and independent of prior index numbers.

* * *

28 Uninhibited criticism of other people's professional efforts, no matter how scrupulous the critic, is bound to be met with hostility in certain quarters. There are always those whom the truth hurts for one reason or another; some devious minds don't even understand the elementary mechanics of truthfulness and will never believe that the child wasn't bribed to call the emperor naked.

We now realize that we should have discounted such hostility and suspicion from the very beginning, as soon as we published the first issue of **The Audio Critic**, instead of responding to every irresponsible attack with indignation and factual arguments, as our charter subscribers will remember. Harry S. Truman, in his admiring reminiscences of General Marshall, observed that the latter had never bothered to answer his detractors. "He wouldn't take the time." We're just beginning to appreciate the full wisdom and integrity of that stance, even if we can't exactly equate ourselves to George C. Marshall in importance. But then our bad-mouthers aren't senators, either, so the ratio remains about the same.

The fact is, in any event, that our ill-wishers have been quite ineffective; our subscriptions show a healthy growth pattern, and the best brains of the audio world appear to be solidly in our camp, as witnessed among others by the distinguished roster of participants in our State of the Art seminar. Every indication is that the teeny-weeny minority of audio hysterics who have inside information about our venality or know a hitherto unsuspected law of physics with which to refute us should from here on be editorially ignored, instead of being elevated from the obscurity they so richly deserve. That will indeed be our basic policy in the future, but from the point of view of our average subscriber there remains a general credibility issue that needs to be addressed.

* * *

29 Suppose we report that preamp A is vastly superior to preamp B or assert, say, that a certain nonnegotiable mathematical truth governs the limitations of subwoofer C. And suppose, no matter how carefully we qualify and document our statements, an underground reviewer or an audio store owner or a manufacturer's chief engineer then declares to all comers that we are 100% wrong, that in fact we are a certain sphincter-bound bodily orifice. How is a busy music-loving doctor or stockbroker to determine at that point who is right? Well, of course, he can't—not without soliciting third and fourth opinions of the highest quality. But we have devised at least a first step toward clearing the air in such cases.

We hereby offer all these critics of **The Audio Critic** the privilege of a tape-recorded debate with the Editor, the uncut transcript of which will be published in our pages. That should separate the mindlessly hostile know-nothings from the responsible objectors, since we don't expect the former to want their lack of substantive arguments documented and publicized. It should also create some lively reading for our subscribers, of possibly even broader credibility than our editorial correspondence ("Box 392"), from which our detractors have been known to cop out because "you can't win when the Editor has the last word." Well, in these transcribed debates anybody can have the last word. All he needs is the courage of his convictions.

* * *

30 Here are the rules we propose. The debate must be restricted in subject matter to assertions already made in **The Audio Critic**. The "challenger" must be an active and recognized practitioner of some sort in the field of audio, a person who has something to lose if it turns out he doesn't know what he is talking about; in other words, not a college sophomore majoring in literature and looking for a lark. The debate can take place in person, at a mutually convenient location, or by telephone. It will be tape-recorded from beginning to end. Either side is allowed to bring consultants, expert witnesses or any other outside support, as long as their number is mutually agreed on in advance. (One or two should be sufficient.) If necessary, a conference call will be arranged. The debate will be limited to 30 minutes, with an overrun of another 15 minutes if absolutely unavoidable. No more than that; we can't allow this feature to take over our entire publication. The exact transcript of the tape will be printed uncut and unedited, in 9-point type. Once the debate has begun, neither side can withdraw; the transcript *will* be printed no matter what. The whole idea is to drive the snipers out of their blinds and into the open.

Other than this standing offer, we propose to take no further notice of potshots taken at us by anyone, anywhere. (Of course, we shall continue to comment on published letters to the Editor, if comments are called for.) So there's the challenge, fellas. If you're so sure you're right and we're wrong, you can now have a field day showing us up before the very eyes of our readers. It's the chance you've been waiting for. Put up or shut up.

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

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

It lasted 15 hours, with eight stalwart men of audio contributing their best thoughts, but when it was over we had barely scratched the surface. Even so—some surface, some scratch! We're publishing the edited transcript in two installments, of which this is the first.

On a clear, sunny Monday in the latter part of the winter, the following eight persons sat around a long table covered with green baize and yellow note pads at the Editor's house and discussed audio from 9 AM until midnight, with only brief interruptions for meals, leg stretching and other calls of nature. In alphabetical order:

Peter Aczel, your Editor and moderator of the seminar.

Mitchell A. Cotter, polymath technologist, possibly the only audio practitioner equally at home in tensor calculus, quantum mechanics, solid-state physics, electronic circuitry, precision tools—you name it—as well as music, now in the business of manufacturing phono system components under his own name.

Julius Futterman, senior member of the group and inventor of the famed Futterman OTL tube amplifier, a man who knows vacuum-tube circuitry like few, if any, others.

A. Stewart Hegeman, second in seniority, a legend among audiophiles for the past 30 years, a pioneer whose engineering roots go back to the old Western Electric days and whose unconventional speakers, amplifiers, preamps and other audio products have been celebrated under names as diverse as Lowther-Hegeman, Brociner, Westminster, Citation, and Hapi.

Dr. Matti Otala, former professor at the University of Oulu, Finland, and now director of the Technical Research Center of Finland, an all-around scientist who has virtually lived inside amplifiers, done extensive psychoacoustic research, and is probably best known in audio circles for his seminal work on TIM.

Andrew S. Rappaport, by far the youngest participant, well known to readers of this publication both for his remarkably original and sonically superior audio designs and for his thought-provoking letters, probably the only authentic whiz kid in audio.

Max Wilcox, our music man and Contributing Editor, producer of Artur Rubinstein's famous albums and innumerable other records at RCA, and now an independent free-lancer experimenting with supersimple, purist recording techniques.

Bruce I. Zayde, mathematician and EE, accomplished organist, trombonist and licensed commercial jet pilot, one of the earliest champions in this country of the Thiele/Small mathematical approach to low-frequency speaker design, and one of the few people we trust on the subject of filter theory and other heavy stuff.

By courtesy of Tandberg of America, Inc., we had a brand-new Tandberg TD 20A four-track tape deck recording in stereo what was being said, at 3¾ IPS on 10½″ reels, through a pair of Tandberg TM6 dynamic microphones. (This is in no way a review or an endorsement of the Tandberg equipment, but we can certainly report that everything worked smoothly and reliably; indeed, we're willing to venture the subjective opinion that the TD 20A is an unalloyed pleasure to use.)

What follows is a very lightly edited transcript of this continuous recording, which was interrupted only during meals and intermissions and takes approximately 10 hours to play in its entirety. Asterisks (***) indicate omitted sections, which are relatively brief and not terribly important.

Here it is, then: possibly the most important contribution of **The Audio Critic** so far to the realistic education and general enlightenment of audio enthusiasts.

EDITOR: Gentlemen, welcome to the first seminar of The Audio Critic. I say first because I'm hoping that similar meetings will be taking place in the future, and just for openers I'd like to say that I'm very proud and happy to have a gathering of people such as this to launch this program because I can't imagine a better group that could have been put together for this purpose. I'm really pleased that all of you could come. It wasn't easy as you know to get everybody together in the same place at the same time. The theme of this discussion is the State of the Art in audio. Now that should be interpreted very broadly, namely: What is possible in the light of present-day knowledge? Are these possibilities being implemented today? Is there a likelihood that they will be implemented in the near future? In other words, how should it be done? And how shouldn't it be done? We'll take the typical audio system piece by piece, and I thought, unless you gentlemen have a better idea, unless you have some sort of objection to the approach I'm about to suggest, I would like to start from the listener's point of view. The listener is faced with a loudspeaker, or a pair of loudspeakers, or a number of loudspeakers, which is the thing that he actually listens to, and we could start with that and trace the signal back to the power amplifier, preamplifier, the phono system and various other sound sources. I thought that even before we do that, so that we don't duplicate observations and efforts and discussions later on, we could lay down some ground rules and talk about the listener himself. And that includes the ear, the listening environment, the air, and whatever else you would like to talk about. Is there anyone here who thinks that this may not be the best approach? Because we're going to be very flexible.
FUTTERMAN: It's as good as any, I suppose.
WILCOX: Why not? Try it.
EDITOR: I suppose the traditional way is to trace the signal from the source.
COTTER: You're going to start with the listener.
EDITOR: I think we should start with the listener because whatever is true of the listener will be true of the whole discussion.
COTTER: Let me mention something which I think is an ultimate distortion that occurs outside of the audio system as traditionally conceived. I encountered this problem when I was doing researches on quad. And I think that it's a serious problem that will affect judgments about what is happening and its quality and character and we have to, I think, account for it in establishing our criteria. The concept I know is around that an audio system is an absolutely transparent sonic channel. But I think there is a problem of human disorientation that occurs when you have, let's say, a perfectly transparent medium, although you don't have the supporting visual sensations. A good example of this would be—let us suppose that you are actually in a concert hall. We are simulating your home living room environment by means of some superior three-dimensional TV thing where there you are sitting in the middle of this concert hall but you're surrounded by this holographic, utterly complete visual simulation of your living room. And I'm doing it backwards deliberately, because you are there in the sonic environment. It's completely transparent. The holographic system is utterly transparent to the sound but you are sitting in your living room; in fact you're not in a concert hall chair at all, you're sitting in your favorite chair in your living room. And there you are with the living room constructed around you. I submit that this is a very disturbing illusion and that you would not hear the concert hall quite as well or quite the way you would want to, because the whole human response is not purely sonic. And there is a tendency I think to think that we are going to recreate a total sensory impression with the sonic illusion. And that's a defective idea.
OTALA: Are you talking about a physical, sensory illusion or a psychological illusion?
COTTER: Perhaps both, Matti, perhaps both, because I'm talking basically I think about a visual context that is all out of keeping with the sonic impression. And I tell a story about my now very large son. When he was very small, being very at home in my lab, and utterly unafraid of anything except one place, the anechoic chamber, where he refused to go under any conditions, accompanied or unaccompanied, because it was so grossly strange. And I think that we have an effect very much like that when we're presented with a very perfect sonic illusion, but the visual stimuli are all out of keeping with it. And, in evaluating the perfection of either the recording/reproducing sonic channel, or the character of the image that's created, I think this becomes a very important factor. Max might comment because, from an aesthetic point of view, I'd be interested in what you feel goes on. I think there may be significant tendencies to compensate for, or overcompensate for, some of these missing ideas.

". . . it has, in my opinion, been clearly demonstrated by various researchers that hearing theory as it stands now is not valid."

WILCOX: I wrote a whole article about this phenomenon in, I think, the second issue. It was about listening without the visual stimulus; in my experience, if you remove some of those visual stimuli, you really hear better. I don't know if you read that article or not. It was about the whole idea that your perception of what you hear is based a great deal on what you're seeing at the same time, too. On the other hand, what you hear is being filtered by what you're looking at and your attention is being diverted by what you're looking at. That's perhaps a whole other subject.
OTALA: Here we have two factors, I believe. The first being, of course, that the complexity of the total image is, as you say, a multisensory matter. But in our psychoacoustic experiments, which we have done during the last five years, it has become very clear and evident that hearing and auditory images are at least fifty percent—probably very much more—psychological. I mean, the problem is not that we would not hear accurately. The problem is what we seemingly make of those crude pieces of mosaic that we hear. We insert the missing details in our mind and not through our ears. And this explains a lot of things which came out in those psychoacoustic experiments. For instance, a very trivial relationship which everybody knows: the more active the musician who was tested, the poorer was his sensory illusion of the sound itself, and the better was his illusion of the musical texture. That's one of the important things. A second important thing which came out was that if we picked people who were basically the extrovert kind of people, interested in many things, they proved to be very much more—up to two decades—more sensitive than a group of introverts. These kinds of things, they seemingly play a very, very important role.
COTTER: How you use the sonic information to construct your sense of awareness.
OTALA: That's true.

* * *

EDITOR: Let me ask this. From the point of view of the audio designer, isn't the task accomplished when an exact replica of the original sound field has been created around the ears of the listener?
RAPPAPORT: The thing is, sometimes that's not good enough.
COTTER: That's the point. I think that is the whole point. That's why I was anxious to have Max perhaps recapitulate, but I think that people who do recording often do exactly what Andy is saying. They correct for the absent visual stimuli.
RAPPAPORT: And also the interesting thing, relating to Matti's first point, is that when you listen, a listener's experience to reproduce an event is really anticipatory. When you sit down with a record and you know what orchestra it is, what piece of music it is, you expect something from that, because you've heard the piece performed before, you've heard the orchestra before, or you've heard a recording on that particular label, or produced by that person. You expect something, and you listen to it, and I think—and I haven't done the psychoacoustic tests that Matti's done—I think you expect to hear that, and even if it isn't there you may hear it anyway. And it depends on the type of listener you

are. I think a musician is more keenly aware of the musical content, and an engineer would be more keenly aware of some of the technical aspects, and that kind of thing.
EDITOR: Does this mean that—let us say that technologists have succeeded in duplicating the original sound field in your listening environment. I'm not suggesting that this is possible today or will even be possible tomorrow. Let us say that there is an exact wave for wave duplicate of the sound field around you, and then for psychological reasons the listener says, "I'm not satisfied. This is not the real thing." That does not imply at that point that the loudspeaker designer or the amplifier designer should go back to the drawing board and do something different. It may suggest manipulations outside that area, but it doesn't suggest that. Would you agree?
COTTER: I'm not sure.
FUTTERMAN: Yes, but I'd like to leave psychoacoustics out for the moment and read something that I think is apropos. This is the IRE Transactions on Audio, March/April, 1961, approximately 18 years ago. The Editor's Corner: "Nothing New in Audio." And I'll read it, it's quite short.

* * *

Editor's Note: *To save space and avoid copyright problems, we'll just summarize the timeless editorial by Marvin Camras that Julius Futterman read into the record. It raises the basic questions about phony "breakthroughs" vs. genuine innovation in audio, about the technical criteria of "perfect realism," and about fooling the ear with doctored sound. Accuracy from source through electronic system to listener is conceded to be desirable, but the obstacles are seen as somewhat unyielding. It could have been written in 1979.*

* * *

FUTTERMAN: And this was eighteen years ago, and I think it's very apropos at the moment.
EDITOR: It certainly is. The thing is, if perfect reproduction in the objective sense is possible, if indeed we can duplicate, we can transport a sound field from here to there, what else is there left for audio technologists? Isn't that what this particular discipline is about? To try to transport an exact sound field from here to there?
COTTER: Not altogether.
OTALA: My opinion is that, yes, it is. If we change the sound field, at least in the main signal channel, then we contaminate it. Let everybody contaminate it in his own equipment, if he so wishes. But let's transmit it to him whichever way you choose—record, or radio, or whatever—in an uncontaminated form. That's one of the important things. But the second point is that we apparently do not know how to do this. I would like to say that we are living in the Stone Age of audio, because we do not know what is relevant in this context. I'll just make one suggestion. Up to the end of the '60s, beginning of the '70s, it was okay if you did the amplitude transmission all right. Well, then came TIM and other new findings, which showed that it is not sufficient alone that you reproduce amplitudes. You have to reproduce the first derivative, too. Now, okay, that is the rate of change type of thing. Right now, we have been pondering for a while whether the second derivative is important and indeed it seemingly is. In mechanics—and remember that we are discussing a mechanical device—we have clearly defined properties of derivatives up to the seventh derivative. What goes beyond that, that's another question, and of course we have diminishing gains when we go further, to more and more planes, or spaces, in this respect. But who knows? The third thing is that it has, in my opinion, been clearly demonstrated by various researchers that hearing theory as it stands now is not valid. It certainly must be changed quite a lot. We have now had several theories of how the ear works: the theory of Cardozo and the theory of Keidel, both completely different than present thinking. They offer some explanations, but most probably—whatever the mechanism may be—some new finding will yield us some kind of hearing theory which is far superior to anything we have done in instrumentation today.
COTTER: I think that's a very important point. But I would like to get a little bit more understanding of what you've said. I don't think you really mean that what fact is known about hearing is invalid. What you're saying is that some of the theories that seek to construct a scheme—a predictive scheme, an analytic scheme—with the facts that are known, have holes, or inadequacies. No one who is doing hearing research, I think, thinks as a total theory.
OTALA: No, no, there is a hearing theory, which is generally accepted. But seemingly it is not valid. Seemingly . . . Let me explain what Keidel says as being the most up-to-date way of hearing that he knows. It's as simple as this: the mechanical device, the ear, is just a pure mechanical device. Forget about it, because its characteristics have practically no relevance in the hearing situation. Practically no relevance meaning that, quite often, we find that hearing is better if you have hearing damage. I mean, the sensory illusion is better. Also, a fact which came clearly out in our psychoacoustic experiments, the worse the ears of the subject were the better he was in pinpointing different deficiencies.
EDITOR: Matti, do you mean even the perception of phenomena that are many dB down?
OTALA: Yes. Especially those. Let me continue then. What Keidel says is that we should probably revise our hearing theory so that sound perception doesn't occur in the amplitude and frequency domain, as we've thought so far, but instead, the upper auditory pathway is composed of three detectors. And he names these as being the transient detector, the vowel detector, and the consonant detector. He shows some experiments which quite clearly show that this could possibly be the case. He said, hearing is an optimized computer, that it was developed by natural choice—the law of survival. And in the early times it was different kinds of transient sounds which were important, because of the fact that these conveyed clues of possible danger. Later, the two detectors evolved for the understanding of speech. And they are especially trimmed, those computer programs, for extracting from a complex signal different patterns which might be considered as vowels, or consonants—I mean, just to understand speech. He doesn't claim that this is the whole picture, but what he says is that this is at least one logical approach to it. Therefore, we probably have to revise our thinking. I don't take any sides in that, not pro nor contra, but I just note that if the present hearing theory is valid then our measurements must be wrong. Or vice versa. We came to the conclusion that in stereophonic music, for instance, using very pure sources, but *musical* sources, we found 0.003% rms TIM distortion being audible. Now that is completely impossible because it is below the hearing threshold of the subjects.
COTTER: No it isn't, because you proved it wasn't. The problem is in how we construct our analogy.
OTALA: Yes, but let me cite some other . . .
HEGEMAN: Was that the only thing you heard?
OTALA: . . . examples as well. We probably are dealing with a very much more complicated beast than we ever thought of. Because it is the brain we are dealing with.
COTTER: Let me say some things. Because I think it's appropriate that we talk about hearing theory, but there are various theories—there are theories that have come and have gone. Anybody who's lived through the last 30 or 40 years of hearing research and followed it knows that we have a great breakthrough occuring by the efforts of Georg von Bekesy. Bekesy's efforts were very much concerned with answering the question of what is this thing, this cochlea, this mechanism—what goes on? Bekesy's early work was done on dead cochleas, relatively fresh dead cochleas, of human beings, that were examined with various mechanical types of studies to reveal the mechanism of the basilar membrane, and the organ of Corti, which are the hair cells that pick up some of this motion. But there was a very substantial error, not in any neglect sense, but a very substantial error that arose because there's a very significant difference in the elastic properties of the basilar membrane and the cochlear process, between live and dead tissues. The significance of it was not really fully worked out until somewhere over the last 10 or 12 years. And it amplifies, it extends the peripheral mechanism. In other words, if you look at the hearing sensory process as an ear auditory process—and there's some question as to whether that's adequate or not—for instance, the whole body responds. I mean, we talk about foot-

tapping and body-shaking and gut-rubbing bass—that's a very important part of the . . .
HEGEMAN: We've had some rather weird discussions over the dinner table, how do you hear, and with what do you hear?
COTTER: You can hear with your liver, too.
OTALA: But let's put it this way. I think that we know quite well what the cochlea does, and we know how the . . .
COTTER: No, I wouldn't agree, Matti. I think we have yet to learn even what the cochlea does. But there is a split in the thinking of the hearing researchers between what the psychologists or the psychoacoustic research people, who come at it from a very different point of view, from the standpoint of sensory apparatus, divided up essentially into what they call peripheral mechanisms and the cerebral mechanisms. And I would greatly concur, and I'll tell you some of the researches I know of and the things we've done. People look at the classical hearing loss curves—in fact they view as some tragedy what's called presbycusis, the progressive loss of hearing with age. It turns out that that's largely a noise exposure problem, and it's viewed with some alarm. But precisely the mechanisms you're talking about seem to be taking place in that hearing acuity—the ability to make the discriminations—seems not to be affected in quite the way you would think. The psychologists understand this very clearly as an adaptive mechanism, and the body, the mind, the whole human apparatus, has its survival capability largely because we don't have just one cylinder. We have a great deal of redundancy.
OTALA: I don't talk about that at all. I'm dividing, engineering-wise, the whole thing in a number of elements. Take the ear. I would say, well, we know reasonably well how the ear functions. What happens next is the transmission of the sensory information to the brain. And it's exactly there where Keidel found out these things. He doesn't make any claims of hearing, of the ear itself; he only says that he has done some neurosurgery and he has pinpointed these sensory responses, he has pinpointed reactions. He says that from the ear, the reactions, the neuron responses are basically those of amplitude and frequency. There's a transformation of the signal in the upper auditory pathway; the neurosystem conveys the responses into the brain. And there he finds three major nodes which react strongly to these phenomena that he describes. And what he says is that apparently the input to the brain is primarily characterized by these three variables, and not frequency and amplitude.
COTTER: You're saying that the people doing the hearing work today don't think in terms that resemble electrical network theory, which is where a lot of the findings and the studies and the interpretation and the experimental construction came from when hearing research started. It was dominated by telephone workers. And we owe a great deal of our background to Bell Laboratories.
FUTTERMAN: Most of what Mitch and Matti have been talking about is a little over my head—or over my ears. I know as far as my own hearing goes, if it came to a choice of listening to some rock music or Beethoven's Ninth, I would prefer the latter. And I hear—maybe not as well as younger people, but I hear. I think this business that we're talking about now could go on for hours.
EDITOR: I'd like to bring it into focus.
ZAYDE: In terms of the network presentation that you were describing, is it basically leaning towards lumped parameter assessments? Because that I think is suspect to some . . .
HEGEMAN: I don't think it is.
ZAYDE: You don't think so, or is that . . .
HEGEMAN: Well, I just quote from Dr. McIntyre's article on string instruments which was published about a year or so ago, in which he comments that nobody knows the psychoacoustics of hearing, and that if you were trying to make yourself a model of the thing, if you would make a model so goddamn complicated it would take ten years for you to associate and translate all the data that you were going to get on the thing. Now this agrees a great deal with what Matti said, where there's an analysis of string tone, as you hear the bite of the bow on it, you hear the impact, you hear the body of it, and the decay. And again, more or less your three divisions of hearing that you were talking about. And while Dr. McIntyre isn't trying to make a model for hearing, he's trying to make a model of a violin. And he admits that it's a very complicated and very difficult thing to put back into a mathematical presentation. Yet dammit, any good listener can hear the difference between a good and bad violin, and a good and bad string tone.

". . . the different measurement methods, which are purely engineering sequences, may have no relevance whatsoever in hearing."

OTALA: Let me inject here one thing. My point for this rather long treatise was a very simple thing. We don't know at present how we are listening, what we hear. Therefore, we're trying to quantify the things with engineering methods, or engineering analyses, like the different variables that we know, and the different measurement methods, which are purely engineering sequences, may have no relevance whatsoever in hearing.

* * *

COTTER: I think the problem is that a lot of the predictive, analytic things that are based on these engineering, electrical system approximations to hearing suggest approaches that lead, I think, to a dead end. And that's what Matti is saying. You can begin to pursue the flatness of frequency response and the minimization of total harmonic distortion and those classical things to a point where they no longer matter, and in fact where perhaps many other things go to hell in the meantime. And you may not be looking at those things at all, or sensing those differences, but your ears will. The problem with an inadequate model is that you're likely to pursue the wrong thing, or you're in a quandary as to how to make improvements. Now Peter's original assignment here was a discussion of the State of the Art and then an approach to what is possible, what may be done, and what directions we might take. So we are saying, I think, all of us, that there are a lot of things wrong with these approaches because they are not based on hearing, on what you hear. Because we don't actually have a model of hearing. Let me give one specific example which is something of a mystery, and it's a mystery of several kinds. There are phenomena called interaural harmonics and interaural beats. When you present diotically—that is in separate, in each ear—different stimuli, the mind, which is the hearing organ of most significance, constructs relationships that represent beats and interaural harmonics, which have no existence in physical reality. There is no cross-linking between the ears; even the bone conduction values are eliminated from this equation. So obviously there are peripheral mechanisms that produce kinds of phenomena that represent interpretations of whatever it is that's being transmitted. There's also the question of just what peripheral processing takes place. The organ of Corti, the basilar membrane and that whole "mechanical" mechanism do things to these time relationships, patterns, which obviously from not Keidel's work but Moeller's, many others . . . pattern recognition is what the whole human apparatus is about. And *differential* pattern recognition. We're all familiar with the fact that we can go into a room, there's a little fan going, a little background noise; you not only hardly notice it upon entering the room, you accept it as the environment. It's there, and you sort of lose track of it. But if it should be turned off, or increased or decreased, you instantly notice it. The nature of the human response is to notice differences. Another thing that Matti mentioned which is very very important is that when you look at the human hearing responses, even as measured by frequency domain and that sort of thing, and you look at these peculiar relationships—the decreasing threshold at lower frequencies, the contraction of the range of loudness at low frequencies, and all of this kind of appearance—one has to ask a very important question that isn't often asked and hasn't been dealt with much but can produce a lot of understanding, and recently has begun to produce understanding. You look at this and you say, why? Why this relationship? What relationship to nature does it have? And as Matti mentioned, there's a number of people doing the research, notably in Holland and a few other places, have begun to look at this as a matched signal, matched filter, matched detector mechanism, very very well adapted to the way things happen in nature. If you go looking for information, you only want to spend your money, as it were, use your mechanism in a place where there really is some information. And this low-frequency loss property and this nonlinear gain thing at low frequencies are an ideally matched relationship if you take the outdoor world in the forest or on the plain as the environment in which you want to perform detection. And the frequency range, or the wave-length range, the time relationship range, of the human being is very much related to his size and his mobility and the distances and range at which threats and events will influence his behavior. And you can see some similarities in adaptive response of the hearing mechanism in the elephant ear and in the smaller animal ears as somewhat related to those variables. And this work is very, very recent. So we've been missing for many many years, for decades, important ideas about the nature of hearing that have to do with the fact that somehow or other it has a value, it has a meaning, that these things are the way they are. And we lose sight of this as long as we stick to these engineering interpretations and then try to translate them into amplifier measurements and so on.

RAPPAPORT: The interesting thing is that everything you're saying may very well be true or it may very well not be true. There are a few interesting points. One is that where the human ear is concerned, as far as our pursuit—namely the pursuit of higher fi—it's totally unimportant in one sense, because the same human being under the same circumstances—and there are some emotional considerations that also enter into this—but the same person in the same sound field will react the same way, whether the sound field is original or reproduced. So in one sense, the human ear and our hearing mechanism cancel out of the equation when we're dealing with translating a musical event, or an aural event, into a reproduced event.

HEGEMAN: That would be great, Andy, except for the fact that everybody has their own subjective interpretation of what they're hearing, so therefore you would still get . . .

RAPPAPORT: For the moment I'm eliminating . . .

COTTER: Andy's talking about an ideal translation as being a substituted sound field. If it was totally, perfectly recreated, then you're the same in both cases.

RAPPAPORT: That's right. Allowing for the same emotional environment, the same subjective environment . . . it is relevant in one sense, because right now we're dealing with imperfections, and the idea is that we can't lose sight of the fact that the only reason we need the ear is in order to determine what's important and how we're going to respond to various imperfections. There are kinds of distortions, I firmly believe, that occur in nature—and you touched on this—that our ear is going to be used to. I think that one of the reasons we can hear distortions in amplifiers through speaker systems, for instance, which have, at least quantitatively, ten times or an order of magnitude greater distortion is because the distortion is different. There are different kinds, and they're things we hear in nature and that kind of thing. I think in discussing the ear it's important to realize that the only reason we need to bring the ear into it is because what we're doing is far from perfect.

OTALA: Let me inject here something that might support you. Showing how wrong—how basically, terribly wrong—we have been doing things in engineering is the simple example of headphones. All the listening in psychoacoustic research has been done using headphones. We tried that, too, and we found out to our great astonishment that hearing, or the distortion threshold—that level of distortion that was audible, at least in TIM studies—was three to five times higher with all the headphones. We tried five different headphones, the best there were, and invariably, we had to inject three to five times more distortion into the signal before it became audible. What we learn from this is simple: that your model of differential pattern recognition is really valid. But somehow that model doesn't work when you have headphones. But see how simple these things are. First we put headphones on our heads, say. We say, okay, the sound is fantastic, isn't it? We don't hear any distortion. Then we say, oh yes, that's because of the fact that headphones of course are very much better transducers, aren't they? Therefore we say the amplifiers are quite okay, and it's only the lousy loudspeakers that create the distortion. Whereas the situation is just the converse. There's nothing wrong with the headphones themselves; they are probably very much better than the loudspeakers. But it's the hearing geometry or differential pattern recognition which is important, and therefore the headphones and the hearing geometry in that sense have a masking effect on imperfections.

EDITOR: Wasn't this the root cause of the incorrectness of the Fletcher-Munson curves originally?

HEGEMAN: Are they incorrect?

OTALA: Well, perhaps . . . they are or aren't. Who knows?

COTTER: They are.

HEGEMAN: Are they? It's impossible to set them up in any kind of listening condition.

EDITOR: The Robinson-Dadson curves show totally different low-frequency contours. Because of the headphone situation.

OTALA: Most probably.

* * *

COTTER: Well, based on the Robinson-Dadson free-field threshold curves and the comments Matti made, one of the facts before us, though, is that you need the head diffraction effect in order to work with the whole hearing mechanism. This data shows that very, very clearly. And the minute you remove that from the sound field, there are gross alterations in the way in which you are hearing.

FUTTERMAN: You mean with headphones you don't hear that?

COTTER: With headphones you don't hear that unless you've contrived a very, very different sort of system than what is used to put music over headphones.

HEGEMAN: I believe that no one really considers the value or the importance of the bone conduction aspects of the hearing mechanism.

COTTER: We've begun to now, in the last five or ten years.

OTALA: Remember, the important thing is really that, so far, there is not a single publication I know of that was done on distortion perception without headphones. They're always done with headphones.

COTTER: There was a good set of work done rather a long time ago by Feldtkeller at the Technische Hochschule in Stuttgart, actually done not by Feldtkeller directly, but done by a man named Gassler, who found, for instance, that in a lot of the classical nonlinear distortions there was a very great difference in the distortion threshold depending upon what the tones were that were presented. And they presented thirds and fifths and things of this sort, and he found another very interesting thing in this classical, traditional way, that there were closed contours to the distortion thresholds. And that the closed contours happened to center on a rather low value of sound field—that is, the centroid of these closed contours was in general in the region of 68 or 72-82 dB. Now when Robinson and Dadson did their work, and they began to assemble all the data on loudness that had been collected, they noticed very significant differences in loudness measurements—and loudness difference measurements. And they invented a sort of pulling function that said that, in effect, there was some optimum level at which you preferred to hear for the maximum acuity and that as things got softer than that or louder than that, you mentally contracted the scale to bring it into that range. It's very interesting and noteworthy that the centroid of that distortion threshold of Gassler's work and the centroid of Robinson's correction function all lie in the same general area of intensity. This shows in effect how very different the hearing function is, since it's these differences that we're working with, than the common data that is presented as in this function. We tend to hear, I think, this temporal pattern of loudness change very differently from the way this flat-network, two-port kind of construction is . . .

HEGEMAN: What I was going to say about that is, you have to be very careful to set your average level in order to make those curves mean anything. If you're off base 10 dB, hell, they don't mean a damn. And incidentally, Mitch, as you were talking about your hearing and so forth, and the low-frequency end, we all know very well that your ear reconstructs the fundamental when it's given a harmonic structure.

FUTTERMAN: I think Peter Walker said it much more simply in his advertising, in the brochure on the Quad speaker. He says it's very important to adjust the volume so that it's right for your listening environment. And that's much more important than how the bass sounds, how the treble . . . In other words, the volume level is most important.

HEGEMAN: I have yet to hear a decent harpsichord record that doesn't sound as if you were paying a visit to a boiler factory and clang, clang, clang, instead of hearing the picked strings that a harpsichord represents.

OTALA: There is one record—Afka Records in Boston.

COTTER: You used that for the tests . . .

EDITOR: Gentlemen, could I try to pull all this together? The way it seems from this discussion is that the relevance of a correct model of the ear from the point of view of the audio designer would be that, if we did have a 100% accurate and relevant model, we could then concentrate on those aspects of design that could satisfy this model and not sweat endlessly over those aspects that are irrelevant to it. Because, as Andy says, the hearing mechanism is the same in Carnegie Hall as it is in this room. From that point of view . . .

FUTTERMAN: But that's why we need the golden ears!

EDITOR: From that point of view there is no difference. But instead of concentrating on 79 different things, if we had a correct model maybe we could zero in on 17 of them and have a perfect system of reproduction.

HEGEMAN: Hey, what's the good spot to sit in Carnegie Hall, after all?

FUTTERMAN: How much can you afford?

OTALA: Agreed on one condition: that is, that the hearing mechanism is not the same in Carnegie Hall and here. Because we just discussed the other extreme, the headphones. That's another extreme of a contracting listening environment, isn't it?

RAPPAPORT: The hearing mechanism is the same; the environment is different.

OTALA: Yes, but we introduce other things which we don't know . . .

COTTER: The minute you remove the head diffraction, I think you've greatly disturbed the aural perception.

RAPPAPORT: It all depends on how you define the mechanism—whether that includes the head diffraction or not.

FUTTERMAN: Is that true with the binaural recording?

COTTER: Well, only if you have a binaural head that translates. I think the essence of what we're saying is, and I think we all agree, that there are grave errors made in extending these very elemental kinds of frequency and loudness or amplitude relationships to the criteria for the system. And that we are probably missing—this was Matti's original point, since there seems to be a very important temporal pattern interpretive mechanism involved—we are missing probably the most important aspects of hearing that are not continuous sound effects. For one thing, all this data on loudness is based on a more or less continuous effect. And the whole nature of hearing is to sense differences and temporal pattern, and we're not looking at those mechanisms that make disturbances of that kind when we use this oversimplified analysis.

EDITOR: Mitch, are you suggesting that a system of audio reproduction is imaginable that is far less perfect, according to certain criteria, than some of the systems we have today, but more perfect in various neglected areas, that would sound more real to us than these other systems?

COTTER: Absolutely. I know for a fact that we can have a tenth and a quarter and even a half percent of certain kinds of what some people consider today to be ugly distortions, that are inaudible, absolutely inaudible, and that other factors which are not looked at are optimized that produce a totally different impression. Matti's point was, that when you read an rms value of something, in the form of TIM as read by classical technique, you are getting rid of time in interpreting, or in giving, that number. Because that is in effect a long-time average, a time average between stimuli.

"You're saying the ear is an oscilloscope, not a spectrum analyzer."

OTALA: 200 milliseconds, though.

HEGEMAN: That's a long time.

COTTER: That's a long time perhaps for hearing. On a short-term basis, perhaps a 1 millisecond or perhaps even a shorter basis, that effect becomes magnified. Just, even in looking at the engineering parameters, to values like 10%. And we are not doing the right kind of thing in interpreting . . . Well, for instance, one of the great tragedies was the ready availability of spectrum analyzers. It has misguided us. It is so convenient to buy a very expensive piece of machinery, perhaps that you've had to write very lengthy justifications for; they cost kilobucks, many kilobucks. Then you get the spectrum analyzer, and you proceed to use it. And it's very easy to see interesting, intriguing kinds of displays. The fact is that a spectrum analyzer inherently wipes out time as a consideration. It is a grand averager.

OTALA: Let me give here a practical joke, which is a true joke, also. I'm a member of the International Electrotechnical Commission Audio Standards Committee. Five years ago, IEC adopted an average loudness curve of program, of normal musical program material, which was used for rating channels and especially loudspeakers. In the last conference in Budapest, one year ago in November, I started to wonder about that curve because it showed a very anomalous low-frequency content. It went something like this, and it extended down to 20 Hz or so with about something like a 10 dB loss. I started to wonder how they ever arrived at this, because it was contrary to my experience that this would really be the case. Well, it turned out that it was done in England, and the results were confirmed in Hungary by two groups of researchers. Both had used a spectrum analyzer connected to normal radio programs. And you know what a spectrum analyzer does. Well, all this garbage down here at the low-frequency end was syllable pauses—the inter-word pauses registered as 2-Hz or 5-Hz components. Of course, there is "boop"—that is, sound and no sound, and that's the way it registers. How often we have this kind of thing! Well, this has now been corrected.

* * *

OTALA: Again, we have exactly the same fallacy. If we connect a spectrum analyzer or any kind of narrow-band system, this low-frequency end comes from there—I mean these syllabic pause frequencies.

COTTER: You're more interested in the higher frequencies than those data suggest because we're using in a sense a resonating device, which is all out of keeping with what the nature of the hearing mechanism is.

OTALA: Right. We are doing temporal hearing and not long-time temporally averaged hearing. That's the basic engineering fallacy.

FUTTERMAN: You just saved me $5800.

EDITOR: You're saying the ear is an oscilloscope, not a spectrum analyzer.

OTALA: Yes. More or less.

COTTER: The ear in fact is—what Matti is saying is that the data are very misleading if you use the spectrum analyzer because the nature even of the peripheral mechanism in hearing is a transversal filter, a traveling wave system which is inherently much more sensitive to the fine structure in time than it is to the long-term thing. And maybe the low-frequency things . . . the liver, literally, is a much more important integrator—the body response—and the liver is one of the biggest and heaviest things, and if you've got bass that'll move your liver then you really are satisfied, you know?

HEGEMAN: Here's how you listen to an organ, Mitch.

COTTER: That's right, that's right.

EDITOR: I think we have to pass on to the next subject, whatever that may be. I think we're all agreed that an understanding of the hearing mechanism is essential in order to zero in on relevant and irrelevant aspects of audio design. But let us postulate that the fabrication of a clean channel is possible—clean from the point of view of satisfying the human

hearing needs totally. Let us say that such a channel exists from microphone diaphragm to loudspeaker diaphragm. In that case, do you feel it's possible to transport a sound field from here to there?
HEGEMAN: Well, you've left out one very very important thing—the environment in which it's going to be reproduced.
COTTER: Well, I think we started talking about that whole problem and I said that there's a subjective disorientation effect. What we proceeded to do in this last hour of discussion is to discard—largely discard—the traditional methods of evaluating the channel, saying in effect that if one got some modicum of some of these amplitude and frequency things in hand, that thereafter further pursuit in that direction was pointless because there were much more important other mechanisms. And Matti keeps pointing to the time domain as the relevant kind of thing that is missed by all the traditional methods, the spectrum analysis and so on. If we could transpose a sound field, we would have achieved some kind of an ideal. There's no question. We're all agreed on that. We still have the subjective disorientation, and I think we accept that it's going to be a disturbing factor. And I asked Max about that point because you try to substitute some enriched stimuli to make up for some of that. But the problem is the design problem: just precisely how do you achieve this transparent channel? If you could make this channel, you would have a certain kind of transposition.
EDITOR: Are we at the point where we can analyze the channel? I'm not sure whether we are. We all agree we need more than one channel; we generally use two channels.
COTTER: A sound field.
HEGEMAN: Do you want stereo, do you want ambiphonic or just which?
EDITOR: The channel may be clean, or at least clean as defined by psychoacoustic criteria, but is the availability of clean channels a guarantee of accurate sound reproduction; in other words, of the accurate transposition of the sound field from here to there?
OTALA: But Peter, isn't that an academic question, because we haven't got that kind of channel, and we will not in the foreseeable future? Because of all the contamination we've got—think about those compressors and Dolbys and everything that's used . . .
HEGEMAN: The engineer is designing a product. He's actually working on probably one of—I use 30 black boxes—it may be only 20 black boxes. You go in between the chain, the way music is made, and where you actually listen to it. One thing I've certainly discovered over the years—two wrongs don't make a right. Any time you try to compensate for something that's going on up here, you've ended up lousing it up. So, from a design standpoint, you basically look at gain, bandwidth, signal-to-noise, and your time domain. If you can work yourself down to the best possible performance you can get out of whatever apparatus you're using, you have come up basically with a successful design. If that's the state of the art, you're lucky. And the state of the art changes daily.
ZAYDE: The contamination that Matti was talking about is quite a bit different, I think, than what a lot of the practitioners have been led to believe using spectrum analysis as a means through which one can determine this contamination level. It could be, and in fact is, as we're discovering, that these aberrations are very different from what we've ever thought them to be. And that we're leaning towards a reality of the time domain as giving us a significant handle on what is going on as opposed to spectrum analysis, when in fact most spectrum analyses only dealt with a portion of the general Fourier transformation, just the amplitude characteristic, and eschews every connection with the time domain. So I think there are some significant points here.
RAPPAPORT: The idea is—getting back to what Peter was getting at before—as Matti was saying we're still in the Stone Age of audio, and we're still dealing with some very basic phenomena such as tonality. Peter is talking about recreating the sound field and I think by that he means to a certain degree the holographic aspects of the performance—being able to pinpoint where everything is. We're still being able to work on how things sound. I think we have to take it one step at a time. I would like to be able to reproduce the sound of an instrument, even the sound of a single instrument, accurately, much less where it is, and how big it is, and that kind of thing. And that's a very very primitive aspect of reproduction.
HEGEMAN: How loud it is, how big it is, is part of the basic inherent quality of that instrument. It has to be played against very close limits.
RAPPAPORT: Well, the bigness determines the tonality to a certain extent.
ZAYDE: Our tools are limited by our own practical experience in determining what is the "mathematical representation" of the hearing mechanism.
RAPPAPORT: The interesting thing is that, with the determination of the composition of the hearing mechanism or its characteristics, we're doing it from two ends. There's the biological end, which is trying to examine the hearing mechanism itself and determining how it works, and then there's our end, which is looking at various phenomena and determining how that relates to the hearing mechanism. And we're coming up with an understanding of the hearing mechanism based on simple empirical knowledge that we've derived from experimentation.
COTTER: Andy's got a good simple case for us.
OTALA: So let's dismiss Peter's approach, I mean the hypothetical approach that *if* we could reproduce the sound pattern as we should . . . We cannot—and in the foreseeable future we will not, simply because of the fact that there will not be that kind of sound sources available. Our present problem in my opinion is not that we would not be able to do that in the long run—after 20 or 50 or 100 years, we certainly will. But the problem will be a very simple one: all those listening tests, for instance, conducted nowadays with present equipment, relatively often, as you know, yield as a result that nobody hears any difference with any component of the system. The reasons are very simple. There's so much contamination that no matter how much you add to that, nobody hears anything.
COTTER: I would like to sound a note of optimism in the midst of all this pessimism. I think that Andy's simple example is a very good touchstone, because many of the contaminating influences act in very simple ways to affect the sound of a single instrument. And what we're missing in the approaches at the present time is dealing with the ordering of the events that that single sound represents. Since it's a certain kind of pure case, let me tell you of an interesting experiment which I think makes it much more optimistic than Matti's 20 or 50 or 100-year projection.
HEGEMAN: I could care less about that.
COTTER: When we were doing these experiments in quadraphony—we were using a four-channel independent system—we discovered very quickly that the least subjectively disturbing, psychoacoustically disorienting kind of sound was in trying to reproduce some small, simple instrument—not a piano, but some very small, very simple instrument that could conceivably be right there in the room.
HEGEMAN: Guitar.
COTTER: Guitar was one we used. The guitar taught me a lot of interesting things about the nature of the problem. The propriety of that sound to the room removed a lot of other spatial, disturbing kinds of ideas and problems. And we then zeroed in on, I think, some of the very important time domain effects that have affected my thinking ever since. One of the things we discovered in the course of that particular piece of work, and which actually had been in my head for a long time, with respect to phonograph records, was that virtually all of the significant distortion processes in the phonograph mechanism, which is still our primary medium, and to a certain extent even in the tape recording processes, were time-disturbing effects—were time-modulating effects—rather than the kind of thing that had been analyzed. In fact, when you look back over the literature, there were striking examples of the analyses having been correctly conceived in the approach but immediately lost as the analyst or author sought to present to his fellows some kind of easy engineering handle, which was more like the spectrum analysis kind of thing. It occurred to me then that we were neglecting that understanding, that kind of representation. Efforts to examine this time relationship and the mechanisms led to a very dif-

ferent approach to what we were doing. And the sonic effect was a remarkable improvement in spite of all these other limitations in the system. I was very heartened, because it made a bigger improvement in this clarity, this ability to transpose, for that simple sound than any of the other traditional kinds of things. I think we may be a lot closer to that accomplishment than we realize. What was also very heartening and very interesting was that in the phonograph record we had a medium that had considerably more capacity than had been appreciated. Because as we made these changes in approach—and I'll talk more about the details as we get into it—it seemed to me there was more there than had been brought out. We've been able to further that end over the years since. But we had maybe by sheer fortune, sheer good luck, gotten into the medium more than we were taking out because our approach to what was to be done, and our understanding of the mechanisms of the distortion, had obscured for us what really was going on. I am more optimistic. I think that we are much closer to the ability to translate the sound in Andy's sense—as a very vital, uncontaminated sound. I've gone around talking about intellectual honesty in assessing the sound, and I have invented the perfect observer in the form of a 4-year old, impudent and irreverent little girl who simply listens and tells you the truth because she has no big investment in a hi-fi system, and she has no particular concern. And it seems to me that such an observer listening to most hi-fi instantly knows that it's not live; Ella's commercial notwithstanding, there's no doubt. It seemed to me that a sort of ideal was that if this little girl ever paused for a moment to reflect, if she had a moment's doubt, we would already be pretty successful. And that the question, is it live or is it recorded, can be examined from the standpoint of different kinds of criteria than we commonly use. For instance, to remove this environmental problem, which Stew I know feels strongly about too, one of the cute things you can do in assessing system quality—system purity, the lack of contamination—is to walk outside the room. How about that? You've done this, I know.

HEGEMAN: Many times.

COTTER: You walk outside the room, and in effect you have transposed your interpretation.

HEGEMAN: You ask yourself, are they playing it there?

COTTER: Yes. Is there a live sound going on in the room? Because I'm not there, I'm outside there. What is it that's arriving at your ears? For one thing, your frequency response in the traditional sense is screwed up beyond belief. It's tortuous. Your amplitude relationships are obviously affected. In fact, as you walk down the hall, you're introducing a lot of attenuation. But you know, dammit, it's very easy to tell whether it's live or recorded in most cases. And if you've got a system that sounds pretty good down the hall, when you walk in the room it still sounds pretty good, only you begin to become influenced by the lack of realism. I think the answer is along the lines of what Matti introduced as the main consideration, which is that the time relationships with which these sounds arrive are obviously much, much more important than frequency response.

HEGEMAN: And they're more real if you get outside the doorway of the room.

COTTER: The reality is improved. And Max, from the standpoint of the reality of the musical performance . . .

WILCOX: What I'm thinking of is that sometime during the day we should get into the differences between software and hardware. Because I'm the only software manufacturer here.

HEGEMAN: I'm a software user, though.

WILCOX: Okay. Just three weeks ago I recorded the Schubert B flat piano sonata with Richard Goode in RCA Studio A. And we had a very good little piano there, CB 409. What's always difficult is to walk out into the room, position yourself in a certain place because the piano sounds differently of course when I'm going out of the control room door; the piano's quite far away and I hear mostly reflected sound, a kind of clangy bright reflected sound which makes certain seats in Carnegie Hall not make a piano sound very good because you get that kind of clangy sound. The closer I get to the piano, the better it sounds up to a point; then if I get too close to the piano, then it doesn't sound well anymore. If I'm leaning over the piano speaking to Richard or to Mr. Rubinstein until a couple of years ago, the piano sounds very bad there actually. To a pianist that's a rather bad place for the piano to be listened to. In any case, what I try to do is go out there and position myself. Now when I come inside the room, even though I'm down to what I think is a vastly superior way of recording the piano than I used in previous years, because it's two rather widely spaced omnidirectional microphones going on to a 2-track tape, I still always come back, and I say, "Ha, don't have it yet, do we?"

". . . some of the difficulty is that many recording producers and engineers never go out into the hall to begin with."

HEGEMAN: That's true.

WILCOX: Now there are a lot of things involved there—you're involving the room, the loudspeakers, the console, and all kinds of things. I'm constantly faced with that difference, and that's many steps closer to the original than most of us. Because then the tape is processed, then the record is made, then it finally gets on to the equipment that you gentlemen either analyze or manufacture or do other things to. For better or for worse, it depends on characters like me to provide you with the source material that you play. And you were talking about distortions of Dolbys and so forth, if you manufacture equipment to make the average phonograph record sound good, you may be doing very strange things, because the average phonograph record, I submit, still is a rather crudely engineered product.

ZAYDE: Your discussion of capturing the piano deals with the piano as an entire entity. But isn't it true that some people deal with the piano as, really, components—they look at the sound board, they look at the strings, and they try to capture different elements of the piano, such that, when you sum all these things together you get "piano".

WILCOX: The ear doesn't hear it that way.

ZAYDE: That's right, that's the whole point. But that's a fallacy in the practice.

WILCOX: My whole approach to recording—which I always approached as a musician—but certainly the techniques were inherited establishment techniques that I learned from great engineers like Lewis Layton, who were great primitive engineers in that they didn't really know why they were doing it but they did a very good job. The Chicago-Reiner recordings are not the product of a sophisticated engineer but the product of a guy who could hear.

EDITOR: The hall helped.

WILCOX: And a great hall, and a great orchestra.

HEGEMAN: Max, as you transfer between the control room and the studio, and so forth, if this is the fiftieth time you've been in there, you do make your own emotional adjustment to the differences in the sound, do you not? The answer is you walk out in the studio, and you hear how the piano sounds, and you have your more or less particular spot where you want to judge how the piano sounds live in the hall . . .

WILCOX: And that's a subjective judgment itself.

HEGEMAN: Forgetting any kind of recording mode, or anything else. That's where the orchestra's playing, and that's . . .

WILCOX: That's the original subjective thing—where do you want to be?

HEGEMAN: Yeah. That sets your guideline. Okay, now you walk back into the studio, and you start scratching your head and so forth and so on, flip mikes, you do this that and the other thing to try and recreate that. But you do have your own format of the differences between that performance and what you hear in the session, or in the control room. And you make your own mental adjustments, or acoustic adjustments and so forth, and say, "Oh, that's going to be all right." You take these disturbances on there and you say, okay if it sounds like this here in the control room it's gonna be all right.

WILCOX: I think some of the difficulty is that many recording producers and engineers never go out into the hall to be-

gin with. They sit in front of their Altec 604's and they produce what my great old engineer Richard Gardner used to call "typical recorded sound." That's all they're trying to do.
OTALA: It isn't uncommon, really. You've probably heard about Svein Erik Borja's—he's a Norwegian broadcast man—his experiments . . .
HEGEMAN: That's a very interesting article. I enjoyed that.
OTALA: He comes out with comparisons that are just fantastic. For instance, he reinforced Bob Ashley's earlier remark that the equalization curve that was found in most records was the JBL studio monitor loudspeaker's inverse frequency response, first of all. Secondly, he recently showed four different recordings, where the nominal input, the recording itself—the 24-track recording—was made in a hall. It was mixed down by the same mixer in various studios using various control rooms.
HEGEMAN: And he came out with four different records.
OTALA: He came out with such incredible differences that when I heard them I took the table and said, hey this is impossible. And he even included the Rosenborg studios in Oslo before alteration and after alteration. And you wouldn't believe the different sound. So that's the rubbish that you are putting out.
HEGEMAN: No, that's where he has to have a great big shovel to shovel the shit.
EDITOR: Who says we need a mixdown?
HEGEMAN: Hey, how about that?
COTTER: In defense of Max . . .
WILCOX: Let me defend myself for a moment. Then you can defend me. Without making this a sounding board before dying, coming clean with life . . . I still was the product of, and was working for, RCA for 17 years, with an establishment kind of engineering approach. In the last few years—and I think my musical instincts were always very good—but being involved in this kind of electronic thinking has changed my approach to making records. Rather than using many, many, many Neumann cardioid microphones, I have now done things like purchase, or let Unitel Television purchase for me, because they live in my apartment, several Schoeps microphones. When we first bought them we bought 10 cardioids and 6 omnis. But I don't ever use the cardioids anymore; I'm gradually trading them off, because I can't stand to listen to cardioid microphones anymore. We recently recorded the Schubert Eighth Symphony, which Peter has in the sound room, with 6 omnidirectional microphones. I also recorded it with a pair of coincidental figure-8's, which despite the scientific purity of the whole idea I didn't like so much . . .
HEGEMAN: You too, eh?
WILCOX: . . . because there was a certain kind of thing that isn't right about it.
HEGEMAN: Dr. Blumlein . . .
WILCOX: I'd like to hear what you people say about that because maybe it's just that I'm not doing it right. Anyway, what I brought this morning was Peter Serkin playing some Chopin variations, Opus 12, recorded on an ATR 100 at 30 IPS, with no Dolby and a very minimal console—and someday I'd like somebody to build me a really fancy minimal console; oh, I doubt, there probably is no talent in this room that could do that—and fed into two Schoeps omnidirectional microphones that were set in a pattern I had established in another hall; I carry this little drawing around with me, and within certain few inches it seems to work. The mike goes here, and that one goes here, and I sort of have it measured off from the toe of the piano; and it works equally well in varying places . . .
HEGEMAN: Top off, or high stick?
WILCOX: Well, it even has a high stick; as a matter of fact, I stole that from Charlie Fisher. I built a stick that makes the lid about this much higher . . .
HEGEMAN: A real high stick?
WILCOX: Yeah. But anyway, there's no mixdown involved, because the record's going to be cut from a 2-track tape. And I agree, I've seen mixing rooms where you start doing all these crazy things. If you don't have it correct at the date, then the chances for transformation of the tape are infinite. And all you're really doing is adding further distortions of perception and . . .
HEGEMAN: What you really do—you could do this geometrically very easily—you set up a pair of mikes; what you're really recording is the sound field and so forth. Now you start playing in the control room, you start taking pieces out of that pattern. It no longer is the rectangle, no longer is the volume that you started with; it has holes, and this, that and the other thing.
ZAYDE: When you do amplitude match—when that's done—what that does necessarily to time domain relationships, it just blows it apart.
HEGEMAN: Well, what do you think these guys are doing? "That's too loud; let's cut that one down a little bit, let's balance out the sound"—they're not doing anything to the time domain except destroying it.
WILCOX: I don't think this is what Peter wants to get into now.
EDITOR: We can get into this later on. It doesn't really matter. The nature of this seminar is such that subjects keep cropping up that will be caught up with later on. That's completely unavoidable. I don't want to constrain this discussion into any straitjacket here.
WILCOX: I just wanted to say that software is a serious problem for you gentlemen, and there are some people in the world who are trying to give better software to people and I'm just one of them—I hope I'm one of them.
COTTER: I started to say "in defense of Max" because actually you've got a very much more difficult problem than the usual audio designer, who can always resort to specs to prove his case. He can launch a sheet which shows a bunch of numbers and proves unquestionably that he has more zilchus factor and less zorkus factor than anything else in existence. Max has to deal with the sound. He has to deal with the music.
OTALA: There's one added comment on records. We're presenting at the AES 62nd Convention in Brussels, in March, a paper on the signal rates of change that we've measured on records. We show the distribution curves, reproduced for quite a lot of records. There's nothing new about the fact that the slew rates, at a 100-watt amplifier output level, point to something like 2 to 4 volts per microsecond, worst case. But what is important there is, seemingly, one thing. For every record we tried—except the Sheffield Lab direct-cut records, which were the only examples that showed an anomalous curve—but all the others seemed to be slew-rate limited in such a way that the signal rates of change measured in those records had an abrupt end. It was like this, and then pop!
HEGEMAN: Topped off?
OTALA: Topped off. There was an end. And this is completely impossible as far as I can understand basic physics.
COTTER: What pickup did you play the records with?
OTALA: MC-20—that's the new Ortofon—very fast.
COTTER: With what type stylus?
OTALA: That has a biradial elliptic, I believe. The importance here is that there's a slew limit, a seemingly very sharp limit. There is a particular limit to every record—the distribution curves just run vertical after that, so that if you try to measure the signal rate of change when it becomes double—the measuring window becomes double—you go about 3 to 4 decades down in the probability. That only points out that we must have an inherent limiting in the process. Where it is . . .
HEGEMAN: You certainly don't want that kind of cutoff in the system.
WILCOX: Where do you think that happens in the chain? These are direct-to-disc records that you're talking about?
OTALA: No, they were all possible records.
WILCOX: No, but I mean the Sheffield Lab ones—were they direct-to-disc records?
OTALA: Yes, they were direct-to-disc. But they were the only exceptions, and even they showed not abrupt . . .
COTTER: Gradual.
RAPPAPORT: Which is what you would expect.
OTALA: Yes. But based on basic physics, it isn't possible that it would be that steep. Acoustical signals do have a distribution of rates of change, certainly, so . . .
COTTER: Did you try measuring acoustical signals, though?
OTALA: No, we didn't in this experiment. But the important factor is, seemingly, that we are already having in the recording studio somewhere, or in the cutter portion of the equipment, something which is slewing, and based on this kind of rapid decrease—first almost flat spectrum and then rapid

decrease—it's also slewing at quite an appreciable time percentage.
EDITOR: Mitch asked whether you tried any acoustical signals. Did you mean just live through the microphone—is that what you meant?
COTTER: Yes. Because we did do some experiments along those lines quite a few years ago with very small microphones and we found interestingly enough that there were cutoffs taking place, that good-sounding and bad-sounding instruments differ in the cutoff.
OTALA: You're talking about frequency cutoffs.
COTTER: No, I mean rates of change. The pressure gradient effects were very important to the sound; and if you think for a moment about what the nature of the basilar membrane process is, if you think in terms of traveling wave and gradient effects, then it would seem as though there ought to be a cutoff in rates of change. That if you got too steep a gradient what you were doing in effect was mechanically and acoustically producing accelerations that are proportional to a power function of the event, and that you would rapidly climb into regions of stress in the mechanism that would excite other undesirable factors. That in fact many a good-sounding violin—and we have Carline's work and others to show that there are interesting differences in the attack and edge effects—that in effect when some people will tell you that a violin doesn't sound very good, it sounds scratchy—that "scratchy," in effect, is excessively high rates of change as compared with the smoothness of the sound of instruments that don't have those effects. So I suspect that there may be in fact an auditorilly determined rate of change that is acceptable, beyond which it is probably painful, since I think there are second-order response mechanisms even at low levels. Some people, for instance, are very, very irritated by the sound of a piece of chalk screeching on a blackboard. And that is in some ways perhaps an example. Because those are very abrupt edges on that kind of sound. Even in reeds, which tend to be square-wave kinds of excitation, a good reed and a bad reed, and a good player and a bad player, differ in the control, in the wetness you might say of the edge on that sound. It's interesting that many of the distortion processes that you find in the time domain that mess up sound add what we sometimes call "fur" in the form of excessive edges. In fact, one of the things you referred to, Matti, in the TIM kind of process, is that a very tiny bit of an edge that rms-es at the double-oh level on a short term basis may be very, very irritating and yet timewise occupy a very small interval. So all these things suggest there may be a naturally desirable slew-rate limit.
HEGEMAN: It's very interesting to listen to a plucked string, which is to me one of the guidelines for good reproduction, and there you hear—it has a short enough time that one of the things that you want to hear, of course, is the attack, the body. But in most, a great deal, of the reproducing equipment you hear the aftertaste, an after-ring when the note stops. I've never tried to do much in the lab measurement on the doggone thing, but a guitar string, a harp string, pizzicato on a stringed instrument, it's what you don't hear—that lack of aftertaste which tells you that your loudspeaker isn't ringing and your wave shapes are good. That gets very significant as part of your listening experience.
ZAYDE: To elaborate on what Mitch said, which is rather interesting, is that brass instruments in particular are capable of sounding incredibly unpleasant. And there is a very interesting effect that takes place when you propagate the signal yourself and feel through your own mechanisms when it becomes unpleasant. There are changes that take place that you can feel. When you get into this excessive rate of change modality, there is a sensation that takes place—at least I speak about when I play, on my lips—you can feel it. This blossoms into this rather unpleasant, edgy sensation. There are some instruments that may have this as a general structure that you cannot dissociate from the overall balance of the tone. For example, if we were to listen here in this room to a Stradivarius and compare it to a Guarnerius del Gesu, the effects would be profoundly different. It's seen that the Stradivarius has this bright, almost edgy quality that *can* become somewhat irritating, depending upon the specific instrument and all this kind of stuff, as opposed to the Guarnerius.

". . . present-day approaches to evaluating the clarity of amplifiers are pointless and reveal nothing."

HEGEMAN: But when you get down into there, you have to figure out who's playing it, whether it's their *[unintelligible]* instrument, and whether they are working to get a tone out that they want to get out of the instrument. A good violinist can make six different instruments sound the same.
ZAYDE: Oh sure, you adapt your playing to the instrument. I'm saying, if we eliminate that and just play the instrument raw, we find that there are some very exciting changes there going on.
HEGEMAN: Generally, I'd consider the Stradivarius a little bit overbright and a little bit hard, but I've heard people play them like you can't believe.
ZAYDE: We can adapt ourselves to this aspect.
RAPPAPORT: It's a feedback mechanism.
ZAYDE: But a healthy one.
RAPPAPORT: That's right. It's good feedback.
EDITOR: This is an interesting consideration. It's possible to sit in an audience and hear unpleasant sounds *live* that are quite reminiscent of unpleasant sounds through a high-fidelity system.
COTTER: Let me mention something to you, some experiments that go in this general direction. As a matter of fact, Max, I think this is scientifically interesting; it's also interesting from a music point of view. It's a kind of example of how we live dangerously, you might say—artistically. There were some studies done that examined the dynamics of artistic performances, and it was found by a group of people working from Bell Labs and by several other researchers, a couple of people in Europe too, that interestingly enough there was a pretty large correlation, pretty strong correlation, between the range of dynamic used and the artistic ranking of the performers. That the inexperienced and less able performer performed an acceptable performance using a smaller range of dynamics, less kind of stress. Now that fact is interesting, but it's more interesting, and made very much more interesting, by some experiments that were done in the development of companders and expanders. Two different systems altogether; in one set of work it was involved with digital; another set of work was involved using an analog group of a kind developed by Bob Grodinsky. In both cases, interestingly enough, live piano, live fiddle, live performances of good artistry—I wouldn't say the greatest, Max, I didn't have access to your level of artist—but really very, very good performances. That in each of these cases you could hardly notice—you began to notice—what were significant downward, compressive effects. It took rather a significant difference to pick up the downward compression. But the least little bit of expansion, like 1 dB per 10 dB, became very quickly painful, irritating. And it suggested that an artist who is a consummate artist knows how to dance just along the edge of a cliff. And that, in effect, great artistry is just the maximum satisfying stress, and then no more. So, I would like to know more about the slew rate in the acoustic reality. And I suspect that artistry is concerned with avoiding excessive accelerations that would impart irritating things—particularly short, edgy things—that the ear and the brain are so adapted to picking up. And that in fact, the "stone wall" effect that you found may lie in artistic considerations rather than in any equipment limitation. If they do, it's very important to know this from the equipment point of view, because maybe we just don't need to do any more than a certain amount because if we were to, then we would encounter harshness and hashiness.
EDITOR: Mitch, do you really think that this natural limiting by the performer would appear as a stone wall on actual measurement instrumentation?
COTTER: Yes, yes.
OTALA: I don't think so.
HEGEMAN: I don't agree with that one

bit.
COTTER: I think this is the control of instruments, something along the line that Bruce is talking about.
OTALA: We have to separate two domains here. First of all, what you're talking about is the acoustically enjoyable domain. That goes for every instrument separately. However, if you take a multi-instrument orchestra, for instance, and you play that on a record, there's a definite possibility that there's a summation of signals in such a fashion that a high rate of change occurs now and then. If that is not reproduced, if the final record played does not show that kind of effects, there's something wrong with it. Just take 10 flutes playing slightly different notes—once a minute at least, their waveforms will orient in such a way that there is a high slew rate, a high rate of change produced.
COTTER: I don't know that that's really true. I think we ought to look at that; I think we need to, because I feel that the ear will go into a problem, also the air itself. It'd be interesting to look back at what you did find and try to determine whether or not that represents a pressure gradient that carries us into a highly nonlinear region or not. I suspect that again we are dancing very close to the limits. I think that when you talk about the propagation of sound, I know we encountered an interesting thing just recently, and we've implemented it, and that is that there appears to be a very significant difference for certain kinds of sounds—and virtually no difference at all in other kinds of sounds—in the absolute phase with which the energy is reproduced.
OTALA: Well, my basic approach here, my way of thinking at least, is that these were multimiked recordings, at least 99% of them, or 95%, or heaven knows exactly . . .
WILCOX: 99, I'm sure.
OTALA: 99. And there, if you have those instruments playing, you add them electrically, there must be some kind of distribution in that . . .
RAPPAPORT: Even if what Mitch is saying about the artistic limitations is correct, then, as you say, at some point in time they have to line up, and also there are other considerations. You would expect to see distortions created by the equipment contributing to very high rates of change, and out-of-band information, that kind of thing. You would expect to see it, even if what Mitch is saying is true about an individual musician.
EDITOR: Do you think we're ready to discuss channels of reproduction at this point, components?

* * *

EDITOR: You may talk about anything you would really be interested in talking about. I just feel that from the point of view of this seminar we should cover components one by one because, let's face it, what our subscribers are interested in—that's components.
RAPPAPORT: You can't cover components one by one, because components interrelate, and in addition to that they all have the same problems.
EDITOR: Of course. Shall we say we should mention them.
RAPPAPORT: Independent of what they are, they all have the same problems.
COTTER: Hear, hear. In fact, I would go further. I would say that Andy's point is very well taken, that in fact you can assemble a whole raft of components that all have these gorgeous specs, and they differ in ways that aren't determined, and they all differ in sound, and they all have similar kinds of problems—and none of them are identified by the traditional methods of measurement. And I'm for one very interested in pursuing further the ideas that the assembled disillusioned souls here have, with respect to both the inadequacy of present methods and the relative importance of the time domain effects.
OTALA: Let me stimulate your thoughts with some examples of components and distortions. Let me cite one which is printed also in our paper, "Correlation of Audio Distortion Measurements." We had an amplifier and we measured it with all the methods that we used. It just went past those methods with very good figures. Then we tried the noise transfer method—you know, putting in pink noise in the frequency range of 10 to 20 kHz and looking what comes down to the range of 0 to 10 kHz. And it showed a very anomalous behavior there; we don't know exactly what causes it. But it is a deep problem. Let's go further. I was recently faced with the problem of different transistors sounding differently. You plugged in transistors and they sounded . . . I know you know that effect. That started to intrigue me so much that I looked very carefully at that. What happened was exactly those things that we have been talking about here. It was a time or phase modulation effect. The simple thing was this: in those transistors the ft varied considerably with current. This affected the sound in such a manner that although the ft was on the order of 15 to 20 MHz, and the stage cutoff frequency around 200 kHz, the first pole, the dominant pole at that circuit shifted back and forth with the signal.
COTTER: To be less abstract, you're saying that the time and the output circuit of the event compared to the input became subject to the value of the current.
OTALA: Yes, you can phrase it that way.
COTTER: It's time modulation, to be less abstract.
OTALA: Time modulation or, if you take sinusoidal signals it's a phase modulation, but all right.
COTTER: But since we're concerned with transients, primarily . . .
OTALA: Yes. All right.
HEGEMAN: Which is a function of top-end bandwidth, right?
OTALA: Right. So the top-end bandwidth went up and down with low-frequency information, and that time- or phase-modulated the high frequency end. Well, that was easy. We later discovered a similar effect in coupling capacitors at the low end. We had a number of coupling capacitors which were electrolytics, and under very special conditions they created an anomaly in the low-frequency response.
HEGEMAN: Which is like a hysteresis loop?
OTALA: You might call it that way, yes. What was the reason then? Quite simple. Namely that they did have a voltage-dependent capacitance, especially at those bias voltages, which were very low. Nothing happened with normal signal components, but since that's an RC network, C in series arm, when we had rumble and record warp signals coming in, they developed an appreciable voltage across the cap, so the capacitance changed, pumped up and down. Consequently, for the low-frequency signals passing this network at the same time, the time or phase was modulated at the rate of the warp. That was very much audible.
COTTER: You modulate mostly the low frequencies, of course, because the high frequencies have no voltage across the capacitor.
OTALA: Yes. Exactly. Below 250 Hz; we didn't detect that at higher frequencies.

* * *

COTTER: Well, I think we've all been looking at these kinds of phenomena because we feel that the present-day approaches to evaluating the clarity of amplifiers are pointless and reveal nothing.
OTALA: Don't put it that way. They are not pointless, but they are not necessarily sufficient.
COTTER: Yeah. If you get below a certain value, that it ceases to have any relevance and that you start to look in other directions. As a matter of fact, one of the things that I've gotten to feel very strongly about is that the mindless pursuit of some of these commonplace values like lower, lower, and lower distortion obtained by more and more and more feedback induces an excess of some of these problems that we are talking about.
HEGEMAN: Creates as many problems as it fixes.
COTTER: Maybe more.
OTALA: This is in fact what I've always been naming the subjective optimum. For instance, let's take a very crude model of your increased feedback—then the static distortions go down and the dynamic distortions go up. Somewhere there's an optimum, and that optimum is particular to any different combination or situation.
HEGEMAN: That's a hardware problem. It's going to be different for every change in hardware that you use.
OTALA: That's true.

* * *

EDITOR: We seem to be on the subject of amplifiers. So let's change our sequence, because everybody's warmed up to the subject, and let's just go on talking about amplifiers. We'll get back to loudspeakers later on.
FUTTERMAN: We never did start on loudspeakers.
EDITOR: I was going to start with loudspeakers . . .
HEGEMAN: That's gonna happen. The

next three days, is that the way it is, Pete? We'll get to loudspeakers in two more?
EDITOR: We may get to them a little bit sooner. But let's continue talking about amplifiers, and let's continue to talk about feedback, because it's one of the really hot issues. I would like to have some more detailed opinions on the subject, particularly as to no feedback vs. optimal feedback vs. too much feedback. I'd like to have everyone's opinion on the subject.
HEGEMAN: As a real old-timer, who grew up before the days of feedback . . .
FUTTERMAN: Me too.
HEGEMAN: Yes, I know, you're right in that class. We lived with some awfully good sound, particularly triodes . . .
EDITOR: You bet.
FUTTERMAN: The Lincoln Walsh . . .
HEGEMAN: Western Electric 300B's. At the time I was working for the Bell System, I used to hear that stuff, and I used to say, Oh God, if only I could make something like that, that sounded like that, in my living room. Okay. Bode, Dr. Black, they came up with feedback; and, commercially, Western Electric went to pentodes, pentodes with feedback. Various other circuits have come up. But they never quite sounded as good as I remember those triodes.
FUTTERMAN: That's nostalgia.
HEGEMAN: I'm sure it's nostalgia.
COTTER: You were at Bell—you recall the caution with which they exercised themselves about 12 dB of feedback, 8 dB of feedback?
HEGEMAN: Oh yes, oh yes.
EDITOR: And it was much less dangerous with the kind of circuitry they were using then than some of the circuitry they use now.
OTALA: There's a good example, however, where feedback just works miracles. You remember how Poulsen found out the "good" tape recorder. It was very simple.
FUTTERMAN: An iron wire.
OTALA: It wasn't the wire. Because the sound was so bad he used feedback, and when he increased feedback, suddenly, like that, there was the sound that was good. And it was fantastically good, better than anything even imaginable at that time. That was the birth of the wire tape recorder. And you know what happened. He had the transformer windings reversed, so the amplifier went into ultrasonic oscillation, and that was the invention of bias.
COTTER: That's marvelous.
HEGEMAN: I think feedback has been used as a cure-all and a catchall. You do a lousy job, and it's supposed to wipe out all the problems. Working with amplifiers—I guess this is 20 years ago, when I was actively doing that kind of thing—one of the things, very simple, you know. You measure your distortion without feedback, you put 20 dB of feedback in, and it should reduce your distortion by 10 times. Suddenly you find it only works maybe 3 times, or 4 times, so you start to look. And the big problem in a distortion circuit is your time delay, your phase delays that are inside your loop. There are amplifiers on the market in which every stage was linearized with its own individual feedback, but that gave an internal bandwidth situation so that in the end loop you could put in some feedback. And frankly, these things sounded cleaner than most everybody else's. You couldn't measure any particular difference in them, but they just sounded better. Every time you increase the feedback you're screwing around with the top end characteristic, you have to put in more phase compensation to keep the thing from oscillating and so forth. It's an area of very little return. So yes, as Mitch said, 8 dB, 10 dB, 12 dB? But something in the art as it was there, turned out to be very, very useful. And it was useful on that thing. This 20, 30 dB feedback kind of thing, you're just building yourself into a hole.
FUTTERMAN: Well, I went through all the things that Stew mentioned, and I wanted to make a perfect amplifier even in the 1950's. I figured the limiting factor was the output transformer.
HEGEMAN: So you threw it out.
FUTTERMAN: So I threw it out. And I don't know if you remember this, Stew, but when you were working with Vic Brociner on the UL-1 on Second Avenue, I brought up one of my prototypes and demonstrated it. And you wanted me to leave it, either Vic or you, and I didn't have any protection at the time, so I didn't. My idea was, if we could eliminate the output transformer we could use feedback. And I worked on that, and it seemed to work. I made amplifiers, and I wrote a paper for the AES in 1954, October, where I described my circuit. I had a pot where you can vary the feedback from nothing to 60 dB—and it was stable, even at 60 dB. It was a class B design. Now there's nothing wrong with feedback; it's very useful. Here is some data on my latest amplifier. Output at clipping: 16 ohms, 115 watts; 8 ohms, 78 watts; 4 ohms, 45 watts. Frequency response, 8 ohm load, 10 watts, 4 Hz to 110 kHz. Open loop frequency, 10 watts with the low-pass filter in, 13 Hz to 22 kHz. With it out, 13 Hz to 24½ kHz. The input low-pass filter, 3 dB point, 110 kHz. Gain, 26 dB. Feedback, 8 ohm load, 37½ dB; 16 ohm load, 50 dB. And I think it sounds good.
HEGEMAN: Probably does.
EDITOR: We know it sounds good. The question is, why does it sound good, and why do other amplifiers with that much feedback sound bad? That's really the issue, isn't it? Both are true: yours sounds good and some of the others sound bad with equal amounts of feedback. Obviously something's different.
HEGEMAN: I think it's pass band inside the loop as being the greatest criterion.
COTTER: Well, you say feedback, but this is a vacuum tube system where the transit time limitations of the vacuum tube don't appear until you talk about phenomena in the several nanoseconds area, and the time dependent delay time, or the current or signal level dependent delay time—these modulations are trivial even in the second order on several nanoseconds basis. So I think what you're dealing with in a vacuum tube system of this kind is something that is grossly —maybe 4, 5, 6 orders of magnitude —different in the delay time change as a function of signal current or thermal history.
RAPPAPORT: Exactly. The biggest problem with feedback is that it tries to take us backwards in time. What it attempts to do is erase a distortion that has already occurred. And you can't do that. The signal occurs in real time, and the reproduction of that signal, the amplification of that signal should occur in real time. Now if you get the transit time, as Mitch said, down to virtually nothing then maybe you are operating in real time.
ZAYDE: There's a well-defined aperture that you can operate within.
EDITOR: Let's talk about that.
COTTER: Anybody who ever tried to park a big boat at a dock knows the importance of the delay time and the rate of response at the rudder. The inexperienced soul can send it out to sea or ram it into the dock in direct proportion to the speed with which he swings the rudder. The cautious soul does an anticipatory sort of thing and inches up to it. And basically what Andy is saying is that if your output—the idea of feedback really is a very simplistic idea. It says, you look at the output, and you compare it with the input, and you correct for the difference. There's a little presumption in there—rather, not a little one, right?
RAPPAPORT: It's a very large presumption. That you don't know what happens in between input and output.
HEGEMAN: When you find a negative time constant that you can put in between the output and the application of the feedback, which will compensate inversely for the delay time through the amplifier, and by God, we've all got it made.
RAPPAPORT: That's exactly right.
OTALA: That is, by the way, not necessarily a difficult thing.
HEGEMAN: I don't have the apparatus on my desk.
EDITOR: Would everyone around this table agree that it's not the number of dB of feedback that must be watched, but the time modulations?
HEGEMAN: Yes.
RAPPAPORT: Actually, it's a combination of both.
OTALA: I don't agree. That's an oversimplification of a complicated matter.

"The biggest problem with feedback is that it tries to take us backwards in time."

First of all, of course, starting from trivial things, people seem to think in terms of feedback being a free entity by itself; you can juggle with it as you wish. This is not true. Firstly, of course, the stability considerations—that is, the compensation and so on—are feedback dependent. This simple reason, as simple as that is, has so far been the reason for about one hundred or so comments on my TIM papers. I've always taken that as being such a trivial thing that it need not even be mentioned. Well, I mentioned it, though, but nevertheless, that's only one thing—the pair, feedback and compensation. The second thing is how do you apply it—because in a given situation, in a given amplifier topology, if you increase feedback you also alter the compensation. But you have to alter something else too, and that's a third variable. And there it breaks. For instance, you have a fixed output level, by virtue of the fact that you are designing a 100-watt or 500-watt amplifier, whatever; you have a fixed input level, and if you increase feedback you have to increase gain somewhere in order to increase the feedback. Because your input and output levels are fixed, and the total gain is fixed. Now how do you do that—that's the crucial point. The problem is that we cannot make a stereotype by saying feedback is bad or good. We have to say how it is applied.

RAPPAPORT: You *can* analyze feedback by itself because, granted, in analyzing feedback as it relates to a specific circuit topology you have to understand everything that's going on. And when you do change the feedback you have to change something else. It could very well be that high-feedback amplifiers have a problem because their open-loop gain has to be so high and their increased distortions occurring from that. However you can analyze feedback in general. A basic feedback system is a feedback system.

OTALA: Assuming that everything else remains constant except that the gain somewhere is linearly increased. The problem is that in a practical situation, when you increase the gain, it's not only the gain that increases . . .

RAPPAPORT: Well of course, you're changing all the other parameters. However, what I'm saying is, forgetting about the fact that feedback is used around a circuit, you can analyze exactly what feedback is. Feedback is more of a philosophy than a technique. You're taking the output signal and returning it to the input.

FUTTERMAN: You're comparing it with the input.

RAPPAPORT: You're comparing it with the input.

FUTTERMAN: And you have an error signal.

RAPPAPORT: Okay, you're creating an error signal. But the problem is that quite independently of what any other circuit parameters are, you're creating an error signal that does not exist in real time.

COTTER: If there's delay. The whole point here is, basically, that if you look at the amplifier—inside of whatever the outside world connections are, which may or may not be feedback—if you look at the amplifier, all an amplifier does basically is to amplify the error signal. Because it doesn't know that it's a feedback amplifier.

FUTTERMAN: It amplifies the true signal and the error signal.

RAPPAPORT: It amplifies the input less the error signal.

COTTER: Well, the error signal is the net difference, however it's taken. The whole point is that an amplifier that has feedback around it does not differ from the amplifier without feedback—as an amplifier. It is simply amplifying the error signal.

RAPPAPORT: Well, it amplifies its input signal, very basically.

ZAYDE: Yes, which is the sum of both.

OTALA: We can simply say that under the nonrealistic assumption that we would change feedback and feedback only—including then, of course, compensation because that's the pair—then we can say that if we are operating in the static domain of the amplifier, or the static operating area of the amplifier, then feedback does good things.

COTTER: You say static because you're saying static and dynamic become synonymous.

HEGEMAN: Without time, you're right.

OTALA: Not really, no; I have to define that for you later, perhaps. Nevertheless, if you're operating in the purely static operational area of the amplifier then . . .

COTTER: The DC response.

OTALA: The DC type of response. That means where you have no signal derivative effects. Then it is purely a good thing. Now, the individual statement, where it is good and where it is bad, depends on where the limit between static and dynamic is, and whether it is inside the band where the amplifier is to be used.

COTTER: Let me ask you a different question which puts the problem in a different light. Let's say that we concern ourselves with the important area of distortions that we discussed a little while ago, which is essentially time modulation effects that arise from the signal or from its recent thermal history. Could we have TIM-like processes without feedback? Let's forget altogether about feedback. And I think the answer is yes, we could have these problems, because we say, in effect, that if there are signal or thermal—which is a sometime integral of the signal—dependent delays, we're going to produce delay modulation effects and we have a problem. Now why in the world is feedback used, to correct for these effects? It seems to me that the basic problem is it doesn't. That's the flaw, the flaw in the thinking.

RAPPAPORT: Once a time modulation effect has occurred, once a distortion like that has happened, you can't take it away.

HEGEMAN: Once you have a modulation you can't linearly correct it.

OTALA: Within certain limits you are right.

COTTER: The thing is you need a predictive kind of thing, or you need something different. The whole point, though, comes back to an interesting thing which Stew and Julius started talking about, and I had some of these experiences too. That is in the beginning, if you look at the early Bell System papers and you look at the approaches to feedback amplifiers, when one went about using feedback in the Bell System carrier amplifier systems, the first thing you did was to linearize the static behavior of the system, which in those days meant vacuum tube considerations, and the static and even the 108-kHz carrier range meant in a sense virtually no time effects. Then you gingerly applied the feedback, because they were very concerned in those days about the order multiplication of distortions, and about the time magnification effects that occurred, in effect, from the fact that, going back to this simplistic idea that the amplifier amplifies the error signal, if you had a lot of feedback from a delayed replica of the signal, you would in effect be moving the position of the thing. As Andy said it tries to jump backwards in time.

OTALA: Let's put it in another perspective still. If you have got a number of devices that you have decided to use, then they have an intrinsic gain-bandwidth product. Now it boils down to the question, how do you divide this gain-bandwidth product—you may use local feedback in every stage. You make some kind of proportions for the stage gains. Then after that you apply an overall feedback. It's more or less a partitioning problem then. This goes back to my first comment; for a given budget, it becomes a question of optimal balancing. When we are talking about these time modulation effects and phase modulation effects, whatever, then I am not completely certain that leaving the overall feedback, for instance, completely out and only applying different, various forms of local feedback would yield, in that respect, an optimum solution. I think that everything applied cautiously is the best . . .

COTTER: Yeah, but let's go back to something even more basic. I tried to bring to focus the idea that if there's time modulation taking place, then however optimistic your DC virtue seems to be, we have to face one basic question. Why use feedback? What are we trying to accomplish when we use feedback? If there are delay modulations, if there is a significant delay time, and the delay is "modulatable," why use feedback?

OTALA: My point was, if we are operating in the static area, then . . .

COTTER: Ah. But this is the problem. In solid state today, what devices are we possessed of where the delay and the delay modulation components are so trivial as to make that assumption . . .

EDITOR: How about MOSFET's?

RAPPAPORT: Even with bipolars, you can create an amplifier—you don't have to use exotic devices—you can create an amplifier with minimal delay modulation effects.

COTTER: If you're looking for them.
RAPPAPORT: If you know what you're looking for, you can create an amplifier with minimal delay modulation effects and in that case—in that amplifier, for instance, you can apply feedback, and some of the gross effects of feedback will be reduced. Because you don't have these delay modulations to begin with; you're dealing with a constant delay, and some of the dynamic distortions created by feedback will be minimized. But there's another side of the coin. There are two factors—one is, if you have time modulation distortion why use feedback? The other is, even if you don't, why use feedback, or why *not* use feedback? Because there are definite reasons why even in that situation you shouldn't use feedback.
HEGEMAN: One thing feedback will do—it will straighten out a nonlinearity in the transfer characteristic.
ZAYDE: Steady-state. That's the whole point.
OTALA: In a static transfer characteristic, not in the dynamic transfer characteristic.
HEGEMAN: Now, what feedback will not do—and this word is thrown around like mad—once you have a modulation, feedback is no good at all. Because modulation is a multiplication and we don't have any good electronic division . . .
COTTER: Demultiplier.
HEGEMAN: Demultiplier, all right, that's a good word. You cannot demultiply something, once there. This is even in our old AM broadcasting system—once modulated, by God, you're dead. You can't unmodulate what you've got.
RAPPAPORT: Especially if it's modulated in a pseudo-random fashion, as occurs in these amplifiers.
OTALA: Stew, you're perfectly right except for one thing. In a well-designed amplifier, these modulations by themselves are so small, that the piecewise linear approach goes. I usually draw this circuit diagram for everybody who says hey, let's use much feedback. I say look, we've got here an amplifier. We go and we measure the output distortion. What is the distortion level that we are likely to find in, say, an 80-dB feedback amplifier. Say it would be 0.01%.
HEGEMAN: Perfectly adequate.
OTALA: Right. Let's then look what's inside the amplifier, because if all our equations are correct the output distortion must be feedback times internal distortion. And what we find is 100%. Well, 100% is impossible. So, what went wrong? Well, there are many possible reasons, but the best reason here would be possibly that the feedback equations do not apply. And that the whole Laplace transform does not apply in this situation. So why?
COTTER: The math is hyperbolic.
OTALA: Yes, but nevertheless the problem mostly seems to be that the circuit was not linear to start with. And the basic, real problem in all these theories is that they assume perfect linearity. Now Stew, you wouldn't make that kind of amplifier—you would linearize it first; so would I. But if we then, in our open-loop characteristic, without feedback—I'm talking about 0.001% open-loop distortion, and say an open-loop bandwidth of say half a megahertz or whatever—then . . .
EDITOR: Then why do you need feedback at all?
OTALA: Well, then I feel at least confident that I may apply a small amount of feedback without penalties.
EDITOR: For what purpose, in that case?
HEGEMAN: If nothing else, Pete, to wash out the variations in the hardware you're going to be using.
OTALA: There are some reasons, yes. One of those for instance is that in a practical amplifier, a power amplifier for instance, I know no other way than moderate feedback to make the closed-loop output impedance low enough.
RAPPAPORT: When do you consider it to be low enough?
OTALA: Say below 0.1 ohm or so; 0.05, something like that.
COTTER: Well, why is that important?
EDITOR: You have a paper that says it's the open-loop output impedance that's important.

"Isn't it true that the situation in which you can apply feedback correctly and fearlessly is the very situation where you hardly need it?"

OTALA: That is it, yes. But there are also some effects which occur with closed-loop output impedance. But nevertheless, I would point out some other factors. Today I'm not afraid of using feedback as such, because it doesn't give me that much of a headache, if it is used moderately. But the problem is, for instance, a typical interface reaction—the Interface Intermodulation Distortion paper, you probably have seen it. That IIM phenomenon is in fact a very good example of something injected into a loudspeaker, coming back from it, propagating via the feedback into the . . .
COTTER: Let's spell that out. We have a loudspeaker. A loudspeaker contains a system that stores energy. If we had a loudspeaker that didn't store any energy we'd have a rather interesting loudspeaker indeed.
HEGEMAN: Almost an adiabatic, huh?
COTTER: Almost. But the fact is that we store energy in a loudspeaker, and . . .
OTALA: We release it backwards.
COTTER: We release it backwards—there are reactions, some comes back, and it comes back at rather variable and different times. In fact it can come back spread out over a whole period of time, much greater than the initial event.
OTALA: We measured about 50% of the energy coming back during the next 50 milliseconds.
COTTER: Which is a hell of a long time compared to the dimensions of most rooms—or the dimensions of time for most musically important events. The basic problem is, what happens to that energy?
OTALA: Well, let me continue then. Firstly of course you try to dissipate it in a physical resistance—the physical resistance being in this case the open-loop output impedance—but if that does not help, then the feedback will take care of it, inject it into the input as an error signal. And we've measured amplifiers which have the signal ratio of the nominal forward signal to the loudspeaker-generated feedback signal of approximately 6 to 9 dB. That means they're almost equal in amplitude, and there may be an intermodulation, certainly, between the primary signal and this loudspeaker-generated, delayed and frequency-transformed version.
COTTER: Perhaps something even 20, 30, 40, 50 milliseconds away.
OTALA: Yup, that's true.
COTTER: And we know that's long enough to be a discreetly different, highly different sort of sound; and these mental time-ordering processes we talked about clearly would recognize even a little teeny bit of some of this.
OTALA: Right, that's true, especially if we have a signal which for instance is decreasing at that very moment so that there's less masking. But the important point here, in my opinion, is that we're quite often mixing our measurement results and our conceptual thinking—feedback by itself in a perfect circuit like an amplifier itself—and describing the bad effects of feedback due to, for instance, time modulation. Although I support time modulation, for heaven's sake—but describing it as being that, whereas it in fact comes from other properties . . .
COTTER: These are some of the things that Andy mentioned when he said that feedback presents some problems that are very different.
RAPPAPORT: Yes, the idea is, let's take your analysis of a feedback amplifier connected to an energy storage device which is going to kick back some of that energy. Now you have a feedback amplifier with a moderate open-loop output impedance, and not all of the energy is going to be dissipated by the open-loop output impedance of the amplifier, and a portion of it is going to be recirculated back effectively, to the input as an error signal. Now what happens if that error signal is then different than that signal needed to properly dissipate the energy present at the output of the amplifier by the time that error signal has passed through the amplifier, which has some finite delay? There's going to be another error signal created, due to the difference between the original error signal and the actual error at that time, recirculated back. And what can happen in certain cases is that an error signal which was not adequately dis-

sipated by the output impedance of the amplifier, that may have lasted a certain length of time, is being continually regenerated through the amplifier, via the feedback loop, until gradually it decays.
OTALA: Well, this is a known phenomenon, but there's only one objection to that. That is that this kind of interface problems are at their worst at low-frequency cone resonances. And they're practically nil above 5 kHz. So the delay—if we've got a feedback amplifier by itself, then the delay inside the loop cannot be that long, otherwise the amplifier would oscillate.
RAPPAPORT: What happens, though—and I honestly haven't done the mathematical analysis necessary to prove this —but what I feel is happening is that the error signal created by the feedback, which is much different than the error signal which the feedback is intentionally creating—in other words, that is the error between the ideal feedback amplifier that is delayless, and the actual feedback amplifier, which can be very short, and its duration is equal to the delay of the amplifier plus the delay of the feedback—creates, in many cases, a regenerative kind of effect which can take a couple of hundred nanoseconds, say, error, and if it is recirculated through the amplifier a hundred times, can make an error signal or a distortion which is clearly audible. And this happens not only from energy being fed to the output of the amplifier by the loudspeaker but it happens because the distortions created in the amplifier itself are being recirculated. I feel this is a very significant effect in feedback amplifiers.
HEGEMAN: We're talking about feedback only in terms of a power amplifier. In a preamplifier circuit, for instance, it's very . . . in the first place, anybody who uses a feedback loop as part of the output of the preamp is I think a little bit out of their minds. So you put a buffer in there so the feedback portion of the circuit is buffered from the device, basically by giving one of Mitch's blank wall interfaces in there—you get rid of almost all of this reaction kind of effect from the load back to the source. That's a whole lot more difficult to do in a power amplifier.
RAPPAPORT: However, the effects of feedback as I said are not entirely due to interface, but they're due to actual distortions created by the amplifier and created by the feedback.
COTTER: It's interesting that the early Bell System papers that discussed feedback discussed precisely this regenerative order multiplication type of problem. The tendency to stretch things out in time and increase the order of the problem. The thing that is very different about the old 300B triode, or the triode amplifiers that Stew referred to . . .
HEGEMAN: I've got a couple of them in the lab. Precious!
COTTER: Yeah, they're really exquisite, precious. But the thing that's interesting about these systems is not only did they share this very low time dispersal, very low delay property, but in effect, you had this terribly inefficient plate resistance of the tube, which in the case of the 300B was a very linear resistor, that is it didn't vary very greatly, but they were quite a large part of the power of the system.
HEGEMAN: The 283's had a plate resistance of about 700 ohms, I believe. And the 300B was a little lower than that, between 400 and 500.
COTTER: 500 to 600 ohms because I know that they did a very nice match to the 600-ohm line circuit which was so popular in the transmission characteristics. The fact is, though, that what you had was an amplifier that could be envisioned analytically as essentially a current source, shunted by a fairly fat resistor, a fairly power-grabbing resistor, in parallel with whatever the load was. So that if this energy, from any energy storage system, whether it was a network or a mechanical loudspeaker, did come back, it didn't meet perhaps a stone wall, but it met a purely non-time-dispersive energy absorber, which did a neat little job of damping it. If you had no feedback on such a system, then it reflected it very little if the damping was decent. It's interesting that in a talk Les Paul gave to the AES many years ago about the early days of recording, electrical recording, he talked about some of these things you mentioned, Stew, and he said that as they kept improving the amplifiers, the sound kept getting worse and worse and worse.
ZAYDE: What's interesting is that the transit times of the devices very largely determine the shape of the envelope that is generated. The historical significance becomes profound as you stretch out these internal transit problems.
COTTER: Or as you increase the amount of feedback.
ZAYDE: Precisely. There's a very close relationship.
EDITOR: Can this be quantified?
RAPPAPORT: If the number of recirculations is fixed, and that is roughly fixed by the network independent of the delay, then the net result is going to be determined only by the delay. And obviously the shorter the delay for a given number of recirculations, the less the audible effect is going to be.
ZAYDE: This is a by-product of the convolution profile completely.
FUTTERMAN: You people all puzzle me. Here I've designed an amplifier with loads of feedback, which Andy says shouldn't sound good, so does Matti . . . Wait a minute, let me finish, let me conclude. In the final analysis, we're interested in the way it sounds in a component system, right? And wait a minute . . . I believe my amplifier sounds good.
HEGEMAN: I know it does.
EDITOR: But an explanation for that was offered a while ago, of why it sounds good and why others that are . . .
OTALA: Let me straighten out one thing first. I've always been called an enemy of feedback—I am not. I've made it clear so far in this meeting already, three times, I believe, that there is an amount of feedback, overall feedback, which can be used in every amplifier and that the amount varies. In your case it might be that you are using just the right amount of feedback, or even less, heaven knows. But let's put it this way—in a given situation, the use of an infinite amount of feedback is as stupid as using no feedback at all.
EDITOR: This is pretty heavy now.
OTALA: Let me continue with another example. Let me explain the typical methodology of listening experiments. Well, we say, this is a high-feedback design; it apparently sounds bad; so why does it sound bad? Well, I recently discovered a unit which did not produce TIM at all, although it was described as producing lots of audible TIM-like distortion. The effect was very simple. It was namely so, that since the poles of the transfer function just moved up and down with current excitation, so when used with a large amount of feedback, its phase margin was going up and down. The frequency response varied, depending on the signal level. Therefore it created very much this kind of time effects, phase modulation or time modulation, whatever you wish. But here the important thing is, once again, that effects like TIM, or this phase margin shifting or whatever, are not related to the basic concept of the feedback itself, but a very poor application of the principles. So let me still say once again that your approach probably is okay if you have taken into account all those problems—you have done a piece of good engineering. There's nothing wrong with feedback itself; we can use tremendous amounts of feedback if it's applied correctly. Now correctly . . .
EDITOR: Isn't it true that the situation in which you can apply it correctly and fearlessly is the very situation where you hardly need it?
OTALA: That's true, yes, I fully agree.
FUTTERMAN: You're all wet—because in my design, the loads, an 8-ohm speaker load, is so mismatched to the output tubes that it isn't even funny. The output tubes would like to see quite a few hundred ohms and here they're seeing only 8 ohms. So naturally I have a lot of open-loop distortion. So the more feedback I use, the lower the distortion. And as I pointed out, the feedback goes up with the load impedance because the gain of the last stage goes up with the higher impedance. And in fact it keeps going up and up.
RAPPAPORT: I want to say something. The idea is, and I don't want to be—this is very difficult, because I don't want to insult you by this, I want you to understand that an earlier version of your amplifier was the first amplifier I ever heard that I liked.
FUTTERMAN: Which one was that?
RAPPAPORT: This was an H-3a or something like that that a friend of mine had five years ago, and it was the first amplifier I ever heard that I liked. And I think you make a fantastic amplifier, but I think your amplifier may well be better than you think it is and better than we realize it is. Because there are various problems occurring from your use of feed-

back. Now you get away with it, because your amplifier has very small delay, due to the fact that you're using vacuum tubes and their transit time is low—you get away with it. However, look at the modulation of feedback. The amount of feedback in your amplifier is greatly determined by the load impedance. Now as the load impedance changes, which it does with a reactive load, your feedback is changing. As the amount of feedback is changing, you're changing the parameters of the circuit as a whole.

FUTTERMAN: Exactly.

RAPPAPORT: And you're creating by that distortions that—I don't think we even realize what they are at this point.

FUTTERMAN: Now wait a minute. We've come to the ultimate point.

HEGEMAN: Well, the pole at the top-end compensation is not swinging around that way. If it's got itself out of that area, then that degree of feedback change is not as significant. As Matti has talked about, these things are kinda on the edge.

ZAYDE: It depends how you see the complex conjugate pair. Okay, so what does compensation do? You're gaining a handle on that, and you're clamping it. You're permitting a specific aperture.

EDITOR: Bruce, you've done some calculations on this. Why don't you tell us about that?

FUTTERMAN: Let's finish this.

RAPPAPORT: The idea is that, I think you're getting away with it—if your amplifier operated in exactly the same way, except built into the amplifier there were a delay of maybe a couple of hundred nanoseconds, something like that, instead of the few nanoseconds that it actually is, or a microsecond, or something like that, and you had the same kind of effect happening with feedback determined by load impedance, also adding of course the regenerative effects that I was discussing before, you would have a tremendous amount of trouble. You get away with it and your amplifier sounds excellent because the delay you're beginning with is very small, so the modulation is unimportant. I wonder exactly how much better the amplifier could be, or if in fact it could be better, if it would work without feedback. As I said, it might be better than we think it is.

FUTTERMAN: It's impossible because of the mismatch I have in the output tubes.

RAPPAPORT: It's a practical impossibility.

COTTER: When you say mismatch, Julius, what you've got is a vacuum-tube type structure which is basically a current source. And it's configured in such a way—in fact your whole patent is based upon maintaining its current-source qualities by having essentially screen grid drive. So that what you're dealing with is a situation that's not altogether unlike what we have in a collector output terminated power transistor amplifier. For instance, if you have complementary power transistors, and you take the output from the collector instead of the usual emitter-follower totem-pole type thing, then you have essentially a current source system. And again, let me bring this back to see what goes on in an amplifier by realizing that once you get inside the amplifier it doesn't know and it doesn't care whether it's got a feedback loop around it or not. It is unmindful of the fact that there's feedback. It's simply handling an error signal, which I think is Andy's point. It's handling an error signal, it's delivering the output. It's a current source. Now how is it going to deal with this reflected energy that comes back? That's really what the problem is.

FUTTERMAN: Well, isn't the proof in the listening, after all?

COTTER: Well, maybe, but we're trying to go a little further and understand it. I'm saying when you have a current source and you're trying to make it damp out the energy that's coming back, you have an interesting problem. The better the current source is, the tougher it is to cope with that reflected power. Because it can't really, in a sense, absorb it. If you had an infinite current source, you'd be very hard put to deal with that reflected energy. In fact, what you would wind up doing is sending it back out toward the loudspeaker. Quite apart from the fact that there's feedback or no feedback.

OTALA: But remember, the thing that saves us here is that it doesn't matter how much energy you put backwards into your tube—it won't, unless it starts arcing, do very much harm there. So you can tolerate that.

". . . I think there has been a relatively mindless pursuit of the bottom and the top of the meter scale . . ."

COTTER: But the fact is that you are dealing with this elliptical or circular load line where you've got current and voltage out of phase. What I'm saying is, that if you look at where the dissipation is taking place, if you really had an infinite impedance output stage, then this stage is actually incapable of absorbing power. And it will not, no matter how many zillion dB of feedback I have, it will not absorb or damp the load. Now one conceivable way to improve the situation would be to take your zillion-dB-feedback infinite-current-source amplifier and hang a 750-ohm load resistor like the 300B across it, and it makes it into a magnificent amplifier.

FUTTERMAN: I'm still working.

HEGEMAN: Incidentally, just for the record—Mitch and I have been talking about the historic triode amplifiers that Western Electric used to make—the distortion measurements on that were about 40 dB on second harmonic; that's 1%, more or less, over the band. The third harmonic ran down around 37 dB, which is, what, 1½% and so forth, and these things sounded *so* good.

FUTTERMAN: In Japan they're building them.

EDITOR: The subject of this seminar is, what is the State of the Art? So in amplifier design, what is the State of the Art? Can you agree on this?

HEGEMAN: We found that 25 years ago, Pete.

EDITOR: Could you define it as an amplifier in which these time modulation effects are reduced to an absolute minimum? Would that be a good way of defining it?

COTTER: Well, absolute minimum, we don't really know at this point how low you have to go to be inaudible.

EDITOR: Bruce, why don't you tell us about these calculations that you made?

ZAYDE: Basically, the similarity in the transfer function behavior or profile of the feedback—shall we say appearance—which is basically an expanded polynomial, to that of the filter theory suggests that there are methods in which we can get a handle on what the aperture is that one can expect to yield relatively low time-dispersal problems in the feedback approach. And it's closely related to such things we discussed, which is the transit times of the devices, etc. Because, to a large extent, when all this is collected together, it describes where the complex conjugate pair of poles are going to be located on a unit circle that describes this whole phenomenon. So by operating with compensation and the rest, we find that with given realistic devices surprisingly little feedback is tolerable using current solid-state devices of typical transit time proportions; but in such cases as Julius's amplifier we're dealing with enormously short transit times . . .

COTTER: And very little time modulation.

ZAYDE: And very little time modulation, right, as a by-product, you can instill much much larger amounts of feedback and still be within the correct aperture.

COTTER: It's sort of tragic in a way, isn't it, though, that—can we agree that vacuum tubes, which were so difficult to use—Julius got rid of the transformers and removed one of the great difficulties and so made a great deal of feedback possible—that in the solid state devices, which make feedback so very easy to use with this wanton abandon with which it is applied, they are the least tolerant of that kind of scheme. So one is driven to do things in pursuit of these traditional distortion and frequency response numbers that in effect carry you further and further into a state of problem. Because what we're evaluating isn't what we're hearing. And we're kind of agreed that time modulation effects are the troublesome and the sound-contributing aspects.

OTALA: I don't know whether we have agreed.

COTTER: Well, have we agreed that it certainly isn't the fact that one has 0.01% distortion and another one has 0.001% distortion?

OTALA: Yes, we have basically agreed

that there are time modulation effects and that they are important. And TIM is one of that kind of effects in fact, so I support you wholeheartedly. However, I would not mix a distortion mechanism, which produces something, with the result of an engineering operation sequence called measurement method, which yields us a number. Because let's take any measurement method that we've got now—they are purely engineering methods and they yield numbers which may have no relevance whatsoever in this world, referred to either the distortion mechanisms or the psychoacoustic result. Therefore I don't subscribe to your saying, well, 1% is not audible in that and that respect and 0.00-something is not audible in the other one . . .
COTTER: I don't know where the borders lie, but they certainly . . .
OTALA: Yeah, but what you are talking about has nothing to do with amplifier design. It's just that you're applying a measurement method which has different sensitivities; and if it has nil sensitivity for a given phenomenon you can't still say that . . .
COTTER: But one of the problems I'm talking about is that I think there has been a relatively mindless pursuit of the bottom and the top of the meter scale, which are engineering techniques, in the belief, in the rather cultist or mythological belief, that somehow or other virtue lies in those particular directions—0.01 dB of flatness and 0.0001% of distortion.
EDITOR: Wouldn't you even go further? If you are shown an amplifier and you are told that it has 0.0008% distortion, and that's all you're told about it, wouldn't your red flag go up immediately?
OTALA: Not necessarily. I would ask who has designed this amplifier, and after that I would say, hey, it's probably okay. Or there's probably something wrong with it.
RAPPAPORT: You can have an amplifier, depending on what the amplifier's designed to do, with 0.0008% harmonic distortion with no feedback and no time modulation effects, either.
EDITOR: Has that been done?
RAPPAPORT: Yes, it has been done.
COTTER: I don't think most people are aware that it has been done or that it can be done.
EDITOR: Let's talk about it.
HEGEMAN: All you need is one of these hi-fi freaks coming in, "Look at this magazine! Isn't that the greatest thing you ever heard? Look at what that number is!" And it sounds like crap.
EDITOR: Let me structure the question a different way. I think Mitch is among those who has heard me phrase a question in this manner. If you were told by a tyrant that in six weeks you had to come up with a better amplifier than anyone has designed so far, or else be shot, what avenue of approach would you take, beginning tonight?
HEGEMAN: Get ready to get shot.
FUTTERMAN: I would make my amplifier, and say to the speaker manufacturers, give me at least 32-ohm voice coils or 32-ohm transformers in the case of electrostatics.
EDITOR: So you feel that the Futterman circuit with a high-impedance load is the ideal amplifier.
FUTTERMAN: Right.
EDITOR: Okay, that's one opinion. Let's have some more.
COTTER: Go around the table. Bruce?
ZAYDE: Basically I think, to recap and extend, that to get a retention away from static conditions and pay attention to such aspects which are very real as this convolution, which involves such things as h of tau convoluted with g of tau—which I don't know if I should go into, what that is . . .
EDITOR: Don't use dirty words around this table, please.
ZAYDE: Okay. But to pay attention to the dynamic condition almost exclusively puts us in the right direction.
COTTER: How would you do it, is what Peter's saying.
ZAYDE: How would I do it. Well, I would basically analyze, starting from the devices that I'm using, which I would optimize on their own without using any other techniques and extend from that point forward.
COTTER: Are you saying you'd take devices that had low inherent time delay?
ZAYDE: Exactly.
COTTER: Would you be inclined to use solid-state devices or vacuum tubes? Given that you had to run, you had to . . .
ZAYDE: That's a leading question, well not leading, but dangerous. I'm familiar with solid-state devices; I would use solid-state.
COTTER: If you had to play it safe, I think is what Peter is saying. You'd use solid-state?
ZAYDE: If I had to play it safe and run really quick I'd use vacuum tubes.
EDITOR: What about MOSFET's? I tried to ask that before but nobody paid attention. Does anybody have any opinions here on MOSFET's?
FUTTERMAN: I worked with them; I worked with the siliconic MOSFET's. Frankly, they went bad on me so quick and they cost so much money . . .
EDITOR: Don't they have very low transit time?
FUTTERMAN: I had to drop it. They're not high-impedance devices, no matter what people say—for DC maybe, not for AC.
RAPPAPORT: The point is, there is a way to get any device to work properly. Bruce's equations and his work show that it doesn't matter if the device has very long or very short transit times. I use bipolars because I'm very comfortable with them and I found a way to get them to work satisfactorily. Julius uses tubes because he's very comfortable with those and he's found a way to make those work very satisfactorily. And I don't think that by digging into the use of MOSFET's or V-FET's or whatever you had, you're going to make, just by using different devices, a substantial improvement. Now they may allow you to build a circuit or a topology that has great advantages over what we currently have, but the devices aren't the answer.
FUTTERMAN: Listen, if I were to try to do it again—and you mentioned in a review that tubes are going to disappear—I have some AT&T shares and I got their quarterly newsletter and it says, "Solid-State Breakthrough. Bell Laboratories has devised a way to almost double the velocity at which electrons, subatomic particles, can race through solid-state circuitry, a development it calls a major advance in solid-state technology. A team of Bell scientists developed an impurity-free highway system for electron particles, a unique approach that could lead to even tinier microcircuits with higher speed and more capacity than any such circuitry available today." I'd get to Bell Labs and get some circuits.
COTTER: I think what they're talking about is gallium arsenide Schottky-barrier FET's. They exactly double the diffusion velocity, and that's a fancy way of saying something we know already. But that's for shareholders and not for scientists and engineers. What Andy's saying, in a sense, is that it's a difference in perspective. You would look at the devices differently, look at their operating conditions differently. Are you saying, to put words in your mouth, you would optimize and minimize the time delay effects? The time delay modulation effects?
RAPPAPORT: Exactly. I said I use bipolars because I'm most comfortable with them—that doesn't mean I use them the way they've always been used. The idea is that in order to use them you have to look at exactly what they do, exactly how they work in a circuit and exactly how you can minimize the time modulation effects.
COTTER: Suppose it was possible to make an amplifier with a device that was some kind of real slow, electrolytic, slow-poke kind of thing, and you put in a signal and it came out half a second later. As long as it came out precisely, exactly half a second later . . .
HEGEMAN: Your internal total time delay makes no difference.
ZAYDE: Right. There *[unintelligible]* your independent variable, which is in a sense the device that you're using.
RAPPAPORT: That's right. A record or a tape is a time delay device.
COTTER: If we had a half a second delay, however, I think Matti and everybody would agree that the use of feedback would be highly unlikely—it gives you nothing but trouble.
ZAYDE: Oh no, no, you don't want to do that.

* * *

RAPPAPORT: In terms of delay, a very gross example of the problems of time delay in a feedback network would be that, if Max is recording in Studio A at RCA, and we have this incredible feedback network that stretches all the way to this house, whereby any error that's created by the recording or playback process is miraculously picked up here and transmitted back to Studio A, where it is transformed into an error signal which Max takes into account when he is recording this—and ideally we have absolutely no error from microphone to loudspeaker—but Peter decides to play this record a month after Max has recorded it, and by the time the signal gets back to Studio A at RCA Max is recording something totally different. We end up with much greater amounts of distortion than we orginally bargained for—which is exactly what occurs in these feedback amplifiers.

FUTTERMAN: I think you're wrong.

ZAYDE: Regardless of whatever you were recording, Max, you wind up now with Charles Ives.

COTTER: I object to that on musical and logical grounds.

OTALA: I would like to object, too, because that was too gross an example.

RAPPAPORT: I said it was gross, but to a certain extent it will and does happen in these amplifiers.

EDITOR: Doesn't all this come down to the fact that it has to be quantified? We cannot discuss feedback qualitatively; we have to discuss it quantitatively. Doesn't it come down to that?

HEGEMAN: I don't think so, because I think the quantitative analysis depends exactly on the circuit's topology, the particular apparatus you're using, and there's no universal solution. There's just some guidelines as to how you can work and what you should be doing.

OTALA: Let me put it this way. Delay is a funny problem because it doesn't roll off the amplitude 6 dB per octave, which is supposed to be the case in a feedback amplifier. Therefore, you cannot go beyond the case of having a, say, minimum 60-degree phase margin. You can apply feedback to that point. Taking that as an engineering rule, as the utmost limit where you can go, then in that case I'm not concerned so much about the delay itself, because when the delay gets larger, that automatically limits the amount of feedback you can use, unless you change compensation. The only problem is that the worst-case situation comes when you have a system which has initially a small delay, say a couple of hundred nanoseconds which is typical in audio amplifiers, and then has a large delay time variation with signal.

COTTER: Maybe another couple of hundred nanoseconds.

OTALA: Yes, wobbling up and down. This is a dangerous situation, and that's easily verified. Have you seen, for instance, an amplifier when you just juggle with it a little bit; put some kind of capacitor at the output, and you see it starting to oscillate at a high frequency during one portion of the wave form. You feed in a low frequency signal and it starts oscillating just there somewhere. At that very point the phase margin went to zero.

COTTER: And recovered.

OTALA: And recovered. That is the most dangerous thing you ever can have.

HEGEMAN: Try that same amplifier that has that little squiggle on the sine wave, drive that to clipping and see what happens. It'll go absolutely wild.

OTALA: Oh yeah, that's another case. What I wanted to say is—the delay. I think you said it already once before. Perhaps you also, I don't recall who said it. It has nothing to do with the delay itself, and the only important thing as the end result is the variation. In both cases, the application of feedback cannot take any kind of extreme proportions because stability considerations will *anyway* limit the amount of feedback to an amount suitable in that particular application.

COTTER: You don't mean to say though, Matti, that just because an amplifier is stable the feedback that is applied is okay.

OTALA: No, I don't say that. You must remember my qualifications. First of all, the qualifications are that all the signals injected into the amplifier cause it to operate in the static domain. This means not necessarily only audio signals but also ultrasonic signals that we get from records and pickups, you know, say typically up to 100 kHz or so. There will be no TIM or rate-of-change type of effects, any of those, in that region. Secondly, that the feedback applied is reasonable; and the reasonability is set by other criteria, and some of them are interface, some are stability criteria—I mean

". . . there are some kinds of benefit to be obtained from feedback, but I feel that with the sorts of systems that are involved, it's too dangerous to apply broadly."

stability both as far as high-frequency stability is concerned as well as, say, gain stability or low-frequency stability, depending on the situation. Third, the important thing is to take into account also in feedback considerations—all the other things like interface intermodulations and dynamic load variations, which we know. I've measured negative momentary impedances from loudspeakers; I can show you some graphs of that. So the amplifier, the whole concept, must cope with those loads, too. That again tends toward increasing the amount of feedback, not taking it completely off. Finally, slew rate is normally, in normal everyday amplifiers, caused by feedback. So if you get a little less feedback you would probably increase the slew rate. But again here, the importance would not be to consider slew rate as volts per microsecond, because the important thing seems to be amperes per microsecond. Those are the important things.

EDITOR: You're saying it's a balancing act in each circuit.

OTALA: It's a balancing act, yes.

FUTTERMAN: I wanted to say something in reply to Andy. I've lost the train of thought, I'll try to recollect it. He said if the delay was very, very long and it would be fed back to Max—was that it?—it wouldn't be good, or words to that effect. Well, just to be a little facetious, by that time it would be positive feedback, and Max would make a better record because of it.

RAPPAPORT: In one sense that's true, but you're talking about what Peter would say about the record that Max had made and quite possibly it would then become positive feedback. That was just a gross example, and then you can shrink the dimensions to begin to understand exactly what's happening in the amplifier.

FUTTERMAN: I just want to continue. Feedback is not only used of course in amplifiers; we have servo mechanisms, and all kinds of things. And our body has negative feedback in various ways, and it's a very useful tool if it's used properly.

HEGEMAN: I use it when I go to an audio show. I just turn my ears way the hell down, put a lot of negative feedback in, and survive.

RAPPAPORT: We all use feedback every day of the week. We have philosophical feedback, psychological feedback, physical feedback, in just everyday living. It has nothing to do with audio or electronics.

ZAYDE: We're not trying to match the ear, though. Like in airplanes, they use a balance control feedback for the control panels, but it's a very different type of sensory mechanism you're appealing to.

EDITOR: We have an interesting situation here, because I know for a fact that both Andy and Mitch are designing circuits without feedback; Julius is designing circuits with lots of feedback, and they sound good; Matti has designed some pretty good-sounding amplifiers and he says it's a balancing act; and I think Bruce is also, at least from his mathematician's point of view, saying it's a balancing act; and I think Stew, you're also saying essentially that it's a balancing act.

HEGEMAN: It is. It's a function of the hardware as to what you can do with the circuit.

EDITOR: So we have two representatives here of the theory of "Why use feedback if you can avoid it?"—or is that unfair?

ZAYDE: I believe in the balancing act, too.

COTTER: I would qualify that, I really believe in the balancing act and I think there are two dimensions of my concern that really don't relate to feedback. And I'm completely in accord with what Matti has said. The difference is that I think in certain situations where you do not have control over the entire system, the interface with the output, that your balancing act leads you to some very narrow possibilities, where the amount of gain of benefit to be obtained from the use of the feedback is marginally significant. And I think that in that case I would abandon it. I think my limitations come about because I choose, I opt to use solid-state devices where the delays are significantly large and the delay modulation effects are significantly small. In that sort of system, if you calculate it out and you find out that your optimal design is 8½ dB of feedback, I don't think there's a whole hell of a lot of advantage. If it gets up to 20, you then have to concern yourself with the interface problems that Matti has talked about, and that too becomes dangerous. So what I chose to do was to take a rather different approach, since I don't know at this point in time just how much of what kinds of time-delay modulation effects produce how much of what kinds of disturbances. I tried to find something that was an "overwhelming" of the problem. And in doing that, I decided I was safer; without knowing these properties of the hearing mechanism, I was safer. I relied in order to prove my case on the iteration method to test whether or not I had a transparent channel. The iteration method is very simply to string up in series a number of the things that I believe to be transparent to see whether or not it's a verifiable or discernible difference between none and n strung in series, believing that with the best-resolving loudspeaker systems, and the best of demanding program material, that if I heard no difference—not an identifiable or consistent difference—then I had something that was at least noncontributory. I think that there are some kinds of benefits to be obtained from feedback, but I feel that with the sorts of systems that are involved, it's too dangerous to apply broadly. If I had complete control over a situation, I might be moved to use some, again as a balancing act. But I think the numbers that I see as viable within the constraints of present kinds of systems are very very much less than what are in common use. I think Matti could agree that to him 26 dB of feedback is a healthy number, and not 60 . . .

OTALA: Something between, say, 15 and 26, or 16 and 26, something like that.

EDITOR: You're talking about power amplifiers?

OTALA: Power amplifiers, yes.

COTTER: Certainly numbers like 40, 50 and 60 dB of feedback on solid-state devices are dangerous to the extreme.

HEGEMAN: Get lost. You can't do it.

FUTTERMAN: Mitch, would you use local feedback, instead of overall feedback?

COTTER: Well, however you apportion it, I think Matti made the point that you've got a finite gain bandwidth product to deal with and you can apportion, but you still wind up with a very serious constraint. Now Julius, I think your advantage comes from being able to apply the feedback technique with considerably greater ease because your delay and your delay modulation process in the vacuum tube system are vastly lower, and you are therefore able to execute forms of solution, kinds of topology that I don't think work in a solid-state system simply because what you need in order to make them work is far more feedback than the delay modulation will permit. Now Stew has designed in both camps, and has a feeling for this too. Would you agree with that kind of a generalization?

HEGEMAN: Yeah, I do think that anything over about 26 dB feedback is really on the edge, and certainly, beyond the edge, or all the way on the edge, if you have an output transformer . . .

COTTER: You're saying solid-state or . . .

FUTTERMAN: Is not an emitter follower 100% feedback?

COTTER: No. Not at all.

FUTTERMAN: Not at all? Why?

HEGEMAN: It's not voltage feedback.

COTTER: An emitter follower and a cathode follower are somewhat similar and the fact is there's always a gain of less than 1, and the equations contain this (1—a) term throughout.

FUTTERMAN: Wait. The output of the cathode follower is subtracted from the input, and the difference . . .

COTTER: There are very few cathode followers where your feedback approaches 40 dB and it's a vacuum tube. You have to have a mu greater than 100 incrementally. And there are very few emitter followers where your feedback is any more than that kind of a number . . .

FUTTERMAN: What number was that?

COTTER: 100.

FUTTERMAN: 100 dB?

COTTER: No, where you're talking about a 40 dB type ratio. As a matter of fact, one of the things that's a characteristic problem about emitter followers and cathode followers is that the minute you introduce any reactive loads, you get into some very serious problems because then the output following becomes—you expand and you modulate the difference term. Because in the negative stroke on a cathode follower, or in the negative stroke on an n-p-n emitter follower, the bass disconnects from the output when the rate gets greater than what you can pull down the output reactance with.

OTALA: My concern about emitter followers is somewhat more directed towards the frequency or rate-of-change effects. Firstly, remember that the emitter capacitance, internal emitter capacitance, in an emitter follower results as an inductance at the output. Now this inductance being current dependent, we get usually the problem of having a phase modulation in the emitter follower. Having a phase modulation with the overall current. This doesn't of course apply to a single sinusoid but it applies to multiple signals, where the low frequency components will phase modulate the high frequency components. That is one thing. The second thing is that an emitter follower having an inductive base impedance and capacitive loading at the output, as is usually the case due to strays and loading effects, is a Colpitts oscillator, and therefore . . .

HEGEMAN: You get a very nice negative resistance on them without too much trouble.

OTALA: Yes, and that means it's oscillatory. Now, it is not that dangerous that it is oscillatory by itself, because anyway you're going to suppress the oscillation. However, it exhibits a phase margin which is less than 90 degrees at high frequencies, and that phase margin is current dependent. Therefore again you get a set of complex conjugate poles, and these poles are sweeping up and down like crazy rabbits when you . . .

COTTER: Signal dependent.

FUTTERMAN: And the emitter base capacitance varies with current. It's nonlinear.

COTTER: Yup. And what we're saying is even in a cathode follower case where the transit time is small, you can still get significant reactive effects. This has been known, was pointed out, admirably, in the MIT/Rad Lab series with respect to pulse amplifiers. You tend to get this asymmetrical charge and discharge characteristic with a cathode follower.

HEGEMAN: Besides which, they're nonlinear as hell.

COTTER: Maybe.

FUTTERMAN: It's a 100% feedback device. All the output is fed back to the input. In a cathode follower.

COTTER: Except for a very important difference, and that is that the transit time is very small; there is not a very large capacitance modulation . . .

FUTTERMAN: In the tube.

COTTER: In the tube. When you get into the higher frequency domain, transit time effects load the input with a real part dependent on the frequency squared, and you've got a whole order of magnitude, many orders of magnitude, difference in the range of time in which time-dependent effects enter. So the realizability of that local 100% feedback —100% means all the available output being fed back to the input—but the gain is still less than 1 intrinsically, which is what makes that thing stable. Matti points out that under certain conditions, like the Colpitts condition, if the associated strays exist you can get a gain greater than 1. And it does actually happen even with cathode followers that you get into instability situations. I would never use a cathode follower as an output circuit for an audio amplifier, where I didn't know what cable, what system I was dealing with or, if I did, I'd put a thousand-ohm resistor in the series with the output or something to protect it.

HEGEMAN: You noticed that, huh?

OTALA: But if you use an emitter follower in an output stage, there are ways to get rid of these things, or at least to decrease them to an acceptable value. They will be there, but there are ways.

RAPPAPORT: That's one of the reasons that a class A output works, because the changes in current are diminished with a Class A output emitter follower, and also the heat developed by the output stage decreases the capacitance and decreases the reactive effects of the emitter-follower circuit to a certain degree.

OTALA: That's one thing. Also with certain circuit tricks you can always say, let's have a situation so that the emitter follower sees a capacitive generator impedance. And that's the most important single thing to make it stable. Furthermore, of course, if you really want to be clever, then you try to stabilize the collector-base capacitance variation effect. But all right, there's a lot of small minor

things like that. I seem to be thinking nowadays that's the only output stage that you can really conveniently use because all the other circuits have even more problems. I don't know whether you agree.
COTTER: To summarize what we're saying, really, there's no basic disagreement between what Julius is saying and what we're saying. Simply that Julius did not choose to try his system with a solid-state output device.
FUTTERMAN: I did.
COTTER: You did and it didn't work, you said; it kept blowing up. And that therefore he uses the vacuum tube system and then is able to use a lot of feedback. Whereas if we stick to solid-state devices then we are facing more time delay, and what we're all saying is it comes down to the same thing. I don't know whether Julius sat down and set out to make low time delay, low time variation as an objective.
FUTTERMAN: No, I didn't know anything about it.
COTTER: So we're talking about a different understanding of how to assess what we've done. And we're coming to an interesting conclusion, which is that the standard specifications do not reflect this . . .
EDITOR: Yes, I was about to say that. We need a spec.
COTTER: We need a spec, we need an approach which assesses this time modulation process in a way that relates somehow or other to what we hear, and in fact it's sort of a strange and undefined realm.
RAPPAPORT: Just for the record, just so that it's clear that I don't agree with Mitch or with Matti, I want to express a different opinion towards feedback. And it's really very simplistic. And that is that in a solid-state design, which is really all that I've been concentrating on—and it's important that it be understood that my thinking comes primarily from solid-state work—that feedback gives you nothing. Although you can set up a series of parameters by which it is a balancing act, as you say, and you can say—and I've talked to Bruce about this—that you can look at certain parameters of the amplifier and you can determine that amount of feedback which will not cause any kind of regeneration or distortion caused by regeneration that is above some audible threshold, I don't see that feedback gives you anything. I don't see that it gives you anything that you can't obtain without feedback, that I can linearize a circuit—I think that any good engineer can linearize a circuit to the point where feedback is made unnecessary.
EDITOR: Your statement assumes knowledge of the audible thresholds of ordinary harmonic and IM distortions.
RAPPAPORT: Yes.
EDITOR: You're saying you know those thresholds . . .
RAPPAPORT: No. I'm not saying I know those thresholds yet. Obviously I'm trying to reduce all distortion to as minimal a value as possible. However, I'm not uncomfortable if, when stressed, a circuit of mine exhibits half a percent harmonic distortion.
COTTER: I agree. And in fact I didn't differ with you.
EDITOR: I basically—in fact I've said so in print—that I basically agree. I just wish somebody could tell me what those thresholds are, because let us say, let us just assume that a quarter percent sounds a lot better than half a percent. Then let's have it.
COTTER: I think I mentioned a rather definitive but a rather old paper by the Feldtkeller/Technische Hochschule bunch, Mr. Gassler, who did a series of studies that go a long way toward telling us that some very much larger numbers than we currently bandy about border on inaudibility. And I really agree with Andy because I said earlier that actually my feeling was that the kind of benefit that you could get, and the magnitude of the improvement—the magnitude of the amount of feedback that you could use when you defined all of this—was so trivial that I didn't think it was worth the candle.
RAPPAPORT: It doesn't give you anything, plus the risk of using feedback is incredibly high.

". . . we've all come to the conclusion that it's in the time-domain department, where there don't exist criteria, that the problems arise."

COTTER: Yeah, well, the risk part I emphasized heavily. And I feel that we do agree, and there's danger in feedback.
HEGEMAN: I would like to cast a vote for feedback, if I might. I think that any good engineer can take a given batch of hardware—transistors, resistors and so forth—and he can make a circuit work, and he can probably pinpoint that up to a maximum type of thing. Now the problem is, fellas, we're going to make hundreds of these things. I can assure you that the advantage of a small amount of feedback is gonna make these things come off the line and off the test bench and off the line, a hell of a lot easier than if you have to sit and adjust each one right on the nose before it goes out.
RAPPAPORT: That's really not the case, because a lot of the problem . . .
HEGEMAN: Andy, it is the case, I'm sorry.
RAPPAPORT: No, because one of the things . . . One of the things that I've found—and I've begun to produce products without feedback in production, although limited production, quantities—is that the feedback masks a lot of effects that I think otherwise would be . . . If we use feedback to mask these effects I think we would not be getting anywhere, because I find that in my designs, which don't use feedback, I have to very carefully match transistors; I have to hand-bias everything; it's a very ticklish process.
HEGEMAN: That's what I'm saying.
RAPPAPORT: Now, the point is that in doing this, I'm coming up with a circuit that is far more linear than it would be if I took transistors out of a batch and threw them into a circuit. Now my question is, if we use feedback we're not noticing the effects that this transistor matching and this very, very careful biasing and control of various parameters have—we're not noticing these in a feedback circuit. Or we're not caring about them. Without the crutch of feedback we *have* to look at these. If we looked at these in a feedback circuit, would it make a difference? I think it would. Even though we've got 8, 10, 20, 40 dB of feedback. If you linearize the open loop further, if you match transistors, if you carefully bias—as I said, if you control all the parameters very carefully in production—you may end up with an even better amplifier. In which case, feedback still doesn't give you anything.
COTTER: Are you talking about power amplifiers?
RAPPAPORT: Power amps, preamps . . .
COTTER: Because most everything I've been talking about—I'm treating the power amp, the severity of the power amp . . .
EDITOR: We'll talk about preamps after lunch.
OTALA: Let me inject one thing. I don't see, really, such strong points; I said it earlier too. We can use local feedback, and that means feedback in one form or another—feedback it *is*, in fact, if you decrease the stage gain, that you're introducing, even if you would decrease the collector load—so why does the gain of a grounded emitter stage decrease? Simply because we've got the r_e which gives you local feedback. Feedback is the basic mechanism of gain adjustment in our cases. Now, what makes the difference between local feedback around one stage, a short feedback system which encompasses, say, two stages or three stages—what's then the difference between that and a big amplifier with overall feedback around ten, fifteen stages? What is the division between nested loops inside the amplifier versus overall or local?
COTTER: There is a difference.
OTALA: There is a difference, yes. I'm just advocating that there is a difference but it is not that dramatic. I would like to say that overall feedback of course has all the known pitfalls; but even that can be used to some extent, and anyway we're using local feedback. You say you don't use feedback at all; well, you use a lot of feedback in local stages. All right, this is a gliding scale.
COTTER: I would use feedback in increasing measure as I went down in power level to a certain point where then I would have to start decreasing the use of feedback. This is very much dependent on the properties of solid-state devices.

The maximum feedback . . .
RAPPAPORT: As you say, you can't control the gain of the stage without feedback. Even if there are no feedback components, the r_e as you say is a feedback component. And there's absolutely no way around that. However you're decreasing and diminishing delay in a local stage, and also you're eliminating feedback around an interface.
OTALA: Right, that's true.
COTTER: Your maximum error is a single quadrature, too.
OTALA: Right. I seem to prefer much local, some two-stage type of feedback, especially nested loops, and with just a slight overall portion . . .
COTTER: Which is what I think we're all saying.
OTALA: That's a balancing effect, isn't it?
EDITOR: We'll listen to that one very carefully, Matti.
FUTTERMAN: I'd like to point out that in my amplifier design there are only two stages. And then I apply overall feedback, so it's a very simple device. That's why I think I can get away with it.
EDITOR: I don't think anyone here is in total disagreement with anyone else.
COTTER: No, I think it's interesting, though, that we've all come to the conclusion that it's in the time domain department, where there don't exist criteria, that the problems arise. And we all fear feedback rather than respect it.
EDITOR: Do we all agree on the need for a standard, or a spec at least, in that area?
RAPPAPORT: We first have to figure out what we're trying to accomplish and then we can settle on specs.

* * *

EDITOR: We've almost concluded power amplifiers. Is there anyone here who feels that something important needs to be said on power amplifiers?
HEGEMAN: I think we beat it to death.
EDITOR: A lot of these things we've said are applicable—certainly the feedback considerations are applicable to preamplifiers. Let's talk about preamplifiers for a little while. Who wants to segue from amplifiers to preamps?
COTTER: I'd like to start in a different place altogether, because a preamplifier has to do a certain kind of job.
EDITOR: All right, but in that case we should first talk about loudspeakers.
COTTER: I don't agree at all. Loudspeakers are the last thing in the system, obviously.
HEGEMAN: That's for 3 AM, Peter.
EDITOR: That's not the scheme that we picked, but I'm perfectly agreeable to take the course that's developing here, rather than anything preconceived.
COTTER: Let me bend the group out of shape a little bit then. Because I have a certain perspective. My perspective is that, in order to make a preamplifier, one must first find out what happens in the phonograph pickup, or at least what happens to start with, let's say at the stylus. If you understand that, then I think you're in a position to at least define better some of what has to be done with the preamplifier.
EDITOR: How does the rest of the group feel about that road map?
HEGEMAN: I thought maybe we could work up into that gently. Consider that the preamp, that we know what's going into the preamp, and handle it from there to get it to the power amp.
RAPPAPORT: The point is, I don't know if we do know what's going into a preamp.
COTTER: That's my point.
OTALA: Well, any way you wish, this is a round egg, you can view it from any angle.
EDITOR: That's true. Well, let's do it that way, whatever we're warmed up to.
HEGEMAN: Well, for instance I'm not so sure that the stylus is the only thing; how about a tape head? Or something like that?
COTTER: The big medium that we all look at is a record.
HEGEMAN: I gave them up years ago.
FUTTERMAN: For smoking?
COTTER: As I recall, you had a lot to do with making some pretty slick records.
HEGEMAN: But they were done on tape, Mitch.
EDITOR: He doesn't listen to them in the vinyl form.
HEGEMAN: Well, that's not true. But I made them on tape.
EDITOR: I do know that he uses a lousy pickup.
COTTER: Most of the world has its music delivered in the form of a vinyl plate, and we ought to talk, I think, about that problem.
HEGEMAN: Are we talking about the State of the Art?
EDITOR: That's obviously what we're going to talk about.
COTTER: I'll talk about it later. Go talk about preamps. As a matter of fact, wait a minute, I've changed my mind. Let's talk about preamps. I'm interested in what comes out of the discussion as an approach to preamps, and then we'll talk about what goes on at the pickup and we'll see whether they track each other.
EDITOR: Let Stew launch that, since he . . .
HEGEMAN: Opened my big mouth, did I?
EDITOR: You broached the subject. Preamps.
HEGEMAN: It's a very simple concept. You have a signal coming in, you want to get a signal out, you put some gain in between that, and in our strange way, either off tape or off discs, we have to equalize that to compensate for prerecorded equalization that's in there.
OTALA: And then we all come up with one conclusion. In a miraculous way, it's all screwed up; I mean the sound.
HEGEMAN: Totally.
FUTTERMAN: I don't know; they sound pretty good to me.
EDITOR: You're designing one yourself, aren't you?
FUTTERMAN: Yeah. In my spare time.
HEGEMAN: He never has any spare time, if I know Julius.
EDITOR: Because he's always late on the delivery of the latest handmade power amps.
FUTTERMAN: And I'm improving it.
EDITOR: Yes. Well, do we all agree that not all preamps sound terrific?
HEGEMAN: Very few.
FUTTERMAN: I didn't know they had a sound.
EDITOR: Well, what might be some of the reasons?
COTTER: It was theorized, and loudly declared to be the case by some people, that the only difference between preamps were equalization errors. And that all the differences between preamps disappeared when they corrected these equalization errors. And I believe them.
FUTTERMAN: You do?
COTTER: I believe the experiments that they performed probably got that kind of result. I happen to disagree with the conclusion and I think I know why.
EDITOR: You mean you don't dispute that they heard what they heard.
COTTER: No, I don't dispute that they heard what they heard, but I do think that I understand why they heard what they heard. It has to do with anomalies in the approach to the definition of the problem. Their conclusion was that frequency response differences was all that they heard that was left. The basic problem I think . . .
HEGEMAN: Possibly.
COTTER: Well no, this was reported, as you know, by in fact a fair number of people who all tended to corroborate.
EDITOR: We don't have to conceal their names. This was reported by a group up in Boston. I'm not aware of that kind of emphatic conclusion by any other group, are you?
COTTER: Other people took their tip from them and began doing some of the same things. I think it's important to see why maybe they got some of these results. For one thing, pickups are interesting creatures, and they can be represented in electrical analog by a pretty good low-pass filter. The characteristic impedances of pickups are shockingly significant low-pass filters.
OTALA: And you're pointing to the fact that they used the Shure V-15.
COTTER: Or others. And the cutoff frequency of these *electrical* generator systems is anywhere from 11 or 12 to 14 or so kHz, and their Q's at cutoff vary from highly undamped to moderately damped. They also have loss—eddy—resistance type effects that are somewhat signal-dependent, with the possibility of some time modulation. But the basic thing that they have is an inherently significant low-pass property. It's always been amusing to me that one company in particular advertised the extreme frequency flatness of their particular pickup design, and they didn't lie; it was true. That was the data you would get if you did run a frequency response curve. They gave the parameters of the pickup which indicated something in the neighborhood of a 10 kHz low-pass condition with a Q of about 1½. So when you got this flat frequency response out of this pickup, the only conclusion left was that they had this enormous resonance in

order to produce this flat response. Which somehow or other nobody seems to have picked up on, maybe partly because of this whole emphasis on frequency response, where it's considered to be okay as long as what you come out with is flat.
EDITOR: Mitch, could I interrupt you for a second? Isn't there—you're referring to the Shure type of moving magnet pickup and others of that ilk—isn't it true that this kind of low-pass filter also has a mechanical pull elsewhere, conceivably, and the end result is a circuit that's a resultant of the electrical and the mechanical low-pass filter? Doesn't it work that way?
COTTER: Yes, you can have a mechanical low-pass filter, you can have all kinds of response, but no way, no matter how wide a band pickup you have, are you going to get significant speed or time effects out of a generator whose basic characteristic is that of a rather significantly low-pass filter condition. It's also important to know, if we talk about pickups—and pickups have as a function the translation of the stylus/groove contact into an electrical signal—just what the scale of relationships is involved there, and how the stylus scanning process is inherently a bandwidth-expanding process. That is to say, if you have a low-pass filter condition in the recording—suppose you really recorded 15 kHz mechanically—the nature of the tracing process is such that one expands the spectrum, expands the speed of the events—let's shy away from spectrum—the speed of the events that are taking place is expanded in the time scale by virtue of the scanning process. There are a couple of other things that are going on. When stereo records are cut, because of the vertical angle there is built into the movement an inherently self-time-modulated effect that's being speeded up and slowed down by the signal itself—and the magnitude is about 30% speed modulation by the signal, that's what the vertical angle condition is approximately in present, modern-day recordings. All of these processes amount to time-domain-operative effects, and they are applied right at the stylus. All other things being equal, they exist right at the stylus. So that one would expect that the output from a pickup that reproduces this is going to be itself a very much time-expanded, speeded-up set of phenomena, which—being a velocity sensor as most of the magnetic pickups are—would further expand its time/speed effects; and that the whole equalization process is in effect a way of kidding yourself at the output about the relative magnitude of the speed effects at the input, and then enormously increasing by a factor of 100 to 1000 your sensitivity to any of the lower frequency intermodulation phenomena that may be produced, since that's what equalization does—it expands your sensitivity to the lower frequencies and suppresses your awareness of the higher frequencies. This seems to me to be a condition that invites more serious problems than many if not most of the other portions of the electronics in the system. It seems to me an *underrated* problem in its consequences.
OTALA: I think that you forgot a couple of other effects that might also be important in that respect. First of all, if you take a pickup which is basically of the moving magnet type, then you have changes in the reluctance of the magnetic portion. Consequently, the inductance changes. Since the inductance changes and the capacitances remain the same, you have a sweeping resonant frequency.
COTTER: We have a time modulation . . .
OTALA: Yes, that's a time modulation effect.
COTTER: Especially since the low-pass condition is already well into the audio band.
OTALA: Yes. The second thing is that your compliance at the stylus suspension is normally highly nonlinear and therefore also the mechanical resonance, which also sits in the audio range, sweeps up and down depending on the actual point where the . . .
COTTER: There are two resonances in fact—one is the low-frequency arm resonance; the other is a midband resonance generally somewhere between 500 and 2000 or 3000 Hz, which is the resonance of the compliance of the pickup with the mass, or the effective inertia, of the system. And that midband resonance is the region of minimum impedance, where the motion imparted has the greatest ease.
OTALA: This means that there are a number of time-modulating effects present in the pickup. This might be one of the reasons why moving coils are so much better because both effects are lower in moving coils.
COTTER: Yes. Let me introduce an idea since we're talking about preamps—and right away we're talking about pickups.
HEGEMAN: I noticed that.
EDITOR: I wonder how that came about.
COTTER: Maybe my original suggestion wasn't such a bad idea. I have a notion which I've described before to others, and it seems effective because what it seems to me occurs when we talk about pickups is a sort of perceptual mistake. Because a pickup is so small an object, there's a tendency to collapse all of the dimensions into something that is in the zone of smallness. In the same way that if we think about the cosmos, and all but the most reasoning astronomer types don't really put any great weight into pools of galaxies, galaxies, interstellar distances, interplanetary distances—these things may differ in scale by enormous ratios but it's all vastness. And it's either vastness or smallness that takes us out of the realm of our normal sense of proportions, sense of reason about the size and shape of things. It seemed to me there was a lot of mistake, a lot of mis-thinking going on, because the scale of relationships in the phonograph-stylus-groove-pickup system wasn't clearly in front of us. So I said, let's conceive that either we shrink ourselves down so we can sit on the edge of a groove and look at it, or we can scale up the groove. And the convenient ratio occurs of 12,000 to 1—in English units, 12,000 to 1 magnification. So that 1-thousandth of an inch becomes 1 foot. And on that scale the present groove-stylus system can be visualized. And of course we then have a groove that's typically about 3 feet wide and about 1½ feet deep, and it's a V-shaped right-angle groove with a 2-foot-something sidewall on it. And our now nude, de minimus diamond, which is now about a 4-mil square shank, 13, 14, 15 to 20 mils long, sits in this groove. So if we scale this up and look at it, what we have is the stylus with a superelliptical ¼-mil knob on the end. Quarter mil means that there's two rounded surfaces sitting in this 3-foot-wide groove that are roughly the curvature of a softball, in radius. And we have a 4-foot-square pole here that's 15 to 20 feet tall—that's our nude diamond. And typically, as we get into it a little later, these two rounded surfaces sink into the surface of the record—these 2-foot sidewalls, 2-foot-plus wide walls—these two small spots sink in something in the neighborhood of ¼ to 1/3 inch. And it's connected to a modern-day kind of cantilever, which is a smallish cantilever, maybe only 12 mils in diameter and 1½-mil, 1-mil wall thickness. So this is connected—this diamond, 4-foot-square, 20 feet tall—is pushed into this tube, this pipe, that's 12 feet or so in diameter and its wall about 1 foot thick, and a typical cantilever of about 250 feet long in this scale. The typical cantilever is 250 feet long, connected to a generator that's about 150 feet off the ground back up in the pickup somewhere. The pickup of course is 800, 900, 1000 feet away, the back end of the pickup. This begins to give you some sense of scale as to what's happening. We have this ¼ or 1/3-inch depression in the groove wall on this "humongous" stylus connected to this great big giant cantilever, and there are copywriters and other true believers who evidently feel that somehow or other that is going to convey with great accuracy and delicacy the undulations of this groove wall that exist down here in this little contact. It's a highly suspect hypothesis when viewed from this scale. Admittedly things like density don't scale, but there is some sense of the relationship here that one gets out of this. In approaching the problem of how to build a preamp, or what goes on with pickups, we also began to look—when I con-

". . . we're talking about preamps—and right away we're talking about pickups."

sidered this kind of analogy at some of the dimensions and some of the effects and asked certain questions—when the pickup is playing a fairly quiet record, this scaled-up pickup is going to undulate for this fairly quiet record something in the neighborhood of a fraction of a millimeter. And that's a noise that we're going to hear as background noise. This thing is sitting in the groove wall, and one of the questions that we asked very early was, how far does this stylus push into this groove wall and how does that relationship vary with force? All the while analyses have been made of phonograph pickups and playback styli—all these analyses until very recent times have all been essentially geometric analyses—tracing distortion, all this kind of thing, are geometrical. And it's important to realize in geometrical analysis that there's no size particularly involved. Geometry is geometry. It could be a giantific thing such as my scaled-up analogy or a little bitty thing, but it's geometry. And we learned about tracing distortion; and we learned about tracking error; and we learned about vertical angle and these things. And the elastic side of the equation, when it was considered that we had a hard, rounded thing pushing into a softer material—when that analysis took place, the analysis was based upon a classical Hertzian indenter equation which seemed to work. It was derived basically again from an idea of analyzing how a hard round thing pushes into a soft thing, again with no particular attention paid to size, because it was a purely classical physics analogy, scaleless. Just a hard round thing pushing into a soft material causes stresses and pushes and whatnot. And that even seemed to work when it came to measuring metals and looking at how certain things behave. So when the analyses occurred, rather recently—Miller, who was one of Hunt's students at Harvard about 20 years ago, did some analyses, a little over 20 years ago—and those analyses showed certain predictions which seemed to agree with some of the data except when you started looking into the details, and then it was found by a number of workers that they didn't agree. There were peculiar differences in the observed results of frequency response, for instance. And it wasn't until the middle 60's that we picked this thing up and started to look at it. We found a very strange relationship. If you draw a curve of the Hertzian equation, which is force versus displacement—and it's convenient to represent a range from about a milligram to 10 grams since all the workers who have done any work have been within this range—and you cover the displacement in two orders of magnitude against these four orders of magnitude, you get a line which is a straight-line affair and looks like that. And the observations that we made and several others have confirmed is a curve that looks like this. Practically vertical from the x axis; practically no change, and, importantly, a very large distance. With a quarter-mil elliptical stylus, it is into the vinyl at 1 milligram something in the neighborhood of 3800 angstroms. And it doesn't change a whole hell of a lot over this range. This theory is the one used by all the analysts, and these are the facts observed by myself and three other observers. Jim White, who was working as Hunt's last graduate student, did his doctoral thesis on it, and that was actually published and is available in the literature. That set of facts is so shockingly at odds with the theories that we are moved to stop and ask a number of very important questions. First of all, obviously, there is something very different than the classical physics situation; and the explanations for this involve the same kind of considerations of short-range molecular forces that give rise to a very nonclassical phenomenon in your teacup. There's a little meniscus at the side. Classical physics says that the cup should have a perfectly flat line of liquid that goes right out to the edge because that is the position of least energy in the cup. Only it climbs the edge. And as a matter of fact, interestingly enough, the meniscus at the edge will be the same irrespective of whether you have a washtub, a teacup, or a test tube full of the same materials in contact with the same materials. That there is in effect something about the nature of small size that implies a range of forces and a kind of effect that has nothing to do with this classical geometrical analysis. In fact this does happen with a very tiny stylus on vinyl and even on other materials. As a matter of fact, if classical physics were to work, then the whole system of playing a record probably shouldn't work. Because we get into the situation of having these enormous tons of force per square inch, and materials should collapse, and all of these things that we have been blithely ignoring, which are obviously not at work to destroy the effort, seem to be responsible for making the thing work. And a lot of other things come out of a consideration of this sort, and in fact lead us to ask a lot of other questions about what's happening at the stylus. Not the least of which obvious thing from this is that certainly, the dynamical forces on the pickup, on the stylus, are moving the stylus a very much smaller dimension than we had surmised from the Hertzian type law, so the effect of tracing the groove is not quite so seriously disturbed by the dynamical forces. You're getting something much more like the groove. However, what is the groove that we're tracing? These facts say that you are always immersed in the groove some considerable distance; there is no such thing as the surface. In fact the surface—what we know to think of as surface physics, in the sense of what we call surfaces in modern physics—is a range no greater than 10 to 100 angstroms, even 100 may be rather generous; 10, 20, 30 angstroms distance is considered a surface. Obviously a stylus is not playing a surface; it's playing a considerable region of subsurface. We were moved to ask what is the signal-to-noise ratio of a record and why? You're not playing the surface, you're playing something subsurface. An analogy was created to explain the signal-to-noise ratio of the playback process, and certain tests were made, and they verified this. If you flip for a moment to consider a tape track, a wide-track 30-IPS magnetic tape, you have a playing head that's a wide-track head, and you have a large gap because at 30 inches per second you don't need a small gap. If you have a ½-mil or a ¾-mil gap you will see most all of the magnetic oxide which is of the order of ¾ of a mil thick, and with your wide-track head you pick up all the particles that exist to make the signal, and you get a very good signal-to-noise ratio. What happens if you play back that wide-track terrific 30-IPS master with a cassette head? A cassette head is a real narrow one, let's say it's a 10-mil-wide track; and of course it's also a very narrow gap, so it's only seeing the surface layer, and it's a little tiny percentage of the whole tape track.

HEGEMAN: Like a 50 microinch gap.

COTTER: A 50 microinch gap and you've got a 10-mil track. Quite obviously, you're going to get a lousy result. Now let's suppose two observers are both nominally playing the 30-IPS tape track. One says, "That's a terrific recording, sensational signal-to-noise ratio." And the other one says, "That recording stinks." Now it's not the recording that they're observing; it is the particular act of playback that they're observing. Now without going very much further in this, because there are a lot of other interesting things that come out of it, let's flip back to our analogy, our scaled-up analogy of this diamond playing the groove. We see immediately that we are with our elliptical stylus or our rounded stylus playing a very tiny percentage in width of this 2-foot-plus wide groove wall—like our cassette head. Also, one can imagine that in effect the gap width, which is somewhat like the aperture, is determined not only by the radius of curvature but also by the amount of depth that you are pushing down into this, or absorbed into, because a gram is not much of a push. The amount of distance that you are going through this sea of vinyl, this is like a volumetric scan. You are, in effect—you have a tape track; you have some thickness involved here. In fact, it is the number of particles that are averaging under the stylus that is very akin to the tape track system, and we're getting a signal-to-noise ratio that is the statistical averaging of both the number and the average distribution of sizes of these clumps. If we push harder, we're going to thicken the tape, in effect. If we push harder, we're going to see somewhat more material. Also, if we do the obvious thing that we do in the tape recording, if we go to a full-track head instead of a cassette head, if we go to a very broad line-contact geometry, we will get the equivalent of a wide-track tape head. So all other things being equal, we should be able to perform an experiment in which we use a very wide line contact head, and we push with a larger force, and we obtain a very much better signal-to-noise ratio. And it is a fact that if we do this we do precisely that—we obtain a very much larger signal-to-noise ratio. We've been able to demonstrate 26 to, in some special cases, as much as 30 dB improvement in signal-to-noise ratio from the playback of the same record that you play with the smaller "cassette head" type of scanning. Now, this raises a lot of interesting other specters wherein people have surmised the nature of the recording process incorrectly. When a lacquer master is made, the lacquer master is examined by being played. We talk about how good the surface is—sheer nonsense, because we're not playing the surface, we're playing the subsurface. In lacquer the penetration is even greater. So when we assess the characteristics of the lacquer master, we are not assessing the thing we are going to determine when we metallically plate it, because it is then the surface that we're going to replicate. When we then replicate the surface and examine it, it's strangely noisier. Well, it's not so strange because we never played the surface. The surface is in fact noisier, and it's also distorted. When we then replicate it in vinyl, and play it, we're not playing the surface either, fortunately, because it's terrible. We're playing this subsurface volume. And so the whole scheme of reality is quite inverted from what the mythology of playback has told us we should do. The mythology says light force—nonsense. The mythology says narrow, rounded little bitty stylus—nonsense. The mythology says record wear will be reduced if we play at a light force—that, too, turns out to be nonsense, because from this scaled-up analogy it's quite apparent that if you want to know what a stylus is doing, you don't ask the pickup generator, you ask the stylus. When we contrived to measure the mechanical impedance of a stylus right at the stylus, we discovered a very interesting thing—that it bore very little resemblance to the output of the pickup, and that it had enormous energy storage capabilities—resonance in effect—large mechanical impedance and reactance at high frequencies. At very high frequencies, reaching out into the hundreds of kHz, where the device became very much like a squeaking chalk, a very beautiful ultrasonic abrader, and we're dealing further with a nonlinear compliance. Quite obviously something that changes its value of compression by a factor of maybe 2 or 3 over a range of 10,000 to 1 in force is a highly nonlinear spring. There is the physical reality, or at least some significant differences between the mythology of playback and the physical reality of what goes on at the stylus, which helps to explain not only why it works but some of the important differences that are going to occur when we look at what the signal is to be reproduced. Now realizing that this is going on, and that the basic geometry, the tracing effects, can be reproduced with great accuracy, we begin to have a picture of what the signal is that's going to be presented to the stylus. These higher frequency effects are going to be applied, as Matti has suggested, to a highly nonlinear reactant system in the moving-field kind of pickup and you've got also other kinds of elastic properties. So the problem of the pickup is that it presents to the preamplifier, if its behavior is correctly represented, a very very much different kind of signal than had been surmised, and that it is able to propagate back to the generator through this "250-foot long pipe" with any semblance of accuracy is an act of sheer fortuitousness rather than any grand design. The scale of things is such that we're really only beginning to get an idea of what the stylus is doing. The bandwidth is greatly expanded over what the signal has and there are many things going on that don't belong in the realm of present definitions.

FUTTERMAN: I'm very much awed by all this. I'm wondering if Edison would have invented the phonograph if he knew.

COTTER: I think it's fair to say that if a pool of physicists got together and looked at the problem recently, and it was proposed to invest millions of dollars in this scheme, it would be completely dismissed as utterly impractical. Because

". . . I'm wondering if Edison would have invented the phonograph if he knew all this . . ."

the reasons it works have nothing to do with the geometry and are based on effects which are very little understood and have only recently begun to yield to any kind of measurement. I think it's fair to say that the only reason we got it working is that people were too dumb to know that they couldn't do it, so they went ahead and they did it.

HEGEMAN: Isn't that the way most things happen?

EDITOR: Mitch, does your analysis mean that these fantasies of say a laser beam tracking a groove—and I mean an analog groove, I'm not talking about the digital recording technique—a laser beam tracking an analog groove without any contact and therefore "no record wear" and all that, would result in a much noisier playback than what we get with a blunderbuss pressing down with 5 grams?

COTTER: We did some experiments. Aside from the limitation of the laser as a scanning process—which has serious limitations as far as its resolution is concerned—there are schemes where playing back "just the surface" have been accomplished. What's apparent from that is that the surface is a distorted wave form due to the dynamics of cutting. When a lacquer is cut, that surface is not an exact replica of the signal because there are these elastic properties that distort it at the surface, and that the physical indentation of the playback stylus, again fortuitously, is just nearly exactly a compensation for this. That force-modulated scanning radius of the cutter is too, Matti, something of a time-dispersive effect, even in the cutting. The playback partially compensates for some of this by having exactly a converting quality to it. So if you play the surface you get two things: you get a horrible noise and you get a distorted signal. So it is, again, a contrary-to-fact piece of mythology that the surface is both good and quiet. But it's neither.

HEGEMAN: Garbage in, garbage out.

EDITOR: Just to zero in on the practical end of it—tracking, attempting to track, at a quarter of a gram or half a gram is nonsense.

COTTER: For two reasons. You reduce the effective thickness, the effective size of the substrate that you're seeing, and each little grobble and bobble of the molecules that you're going to encounter is that much more effective in moving the position of the stylus.

EDITOR: Does that mean then that two-thirds of the phono cartridge advertising that we see in the magazines today is basically nonsense?

COTTER: Just like I think we destroyed the advertising of amplifiers a little while earlier. This is a good place to begin to understand the problem of pickups.

FUTTERMAN: I found that out in practice. The heavier I usually made the cartridge, the better it sounds.

EDITOR: Most audiophiles have found that out in a purely pragmatic way. They heard distorted sound, they increased the stylus pressure . . .

FUTTERMAN: And it went away.

EDITOR: And it went away. Nevertheless, they're in an agony of apprehension about the damage that they're causing to their records.

ZAYDE: Absolutely. Most people believe the lighter the better. And they really need to let go of that false thinking. Because they really don't understand the quantum mechanical aspects that you just spoke about.

OTALA: There's one effect, however, which gets worse when you increase the pressure. That is with that viscous type of model of yours, when you increase the needle pressure, the apparent mass of the stylus tip becomes larger and therefore it also becomes force dependent.

COTTER: The mass of the stylus?

OTALA: The effective mass of the stylus. Because part of the viscous material, the vinyl, belongs in the equation to the mass, the effective mass, of the stylus tip.

COTTER: While there's a partial truth to that, the fact is that what the data show is that that effective mass contribution at the interface is there; it isn't a very strong variable of force. It doesn't change greatly with the force. It's almost inescapable.

OTALA: What I'm saying is that at the high accelerations you are using, then at high pressure levels the added mass contribution becomes variable.

COTTER: No. It's not significant. The data show that there's a very, very small change—much less in fact than you would expect from a Hertzian law. Actually, the material behaves, for the volumes and the frequencies involved, except at very high ultrasonic frequencies, essentially as a stiffness component. What does change with force is the real part, because this nonlinear capacitance is still nonlinear but it doesn't dissipate any energy. What does change is the mechanical resistance, which absorbs more and more energy as you go up in force.
OTALA: You're saying that the vinyl acts as a limited-slip oil.
COTTER: It's a highly non-Newtonian system. As a matter of fact, you improve the ultrasonic damping at higher forces in any given system.
OTALA: Yes, you improve the damping, but you cause variation, more variation, in the mass for lower frequency components.
COTTER: No, the increase in apparent mass is not a very large factor at all, because the change in distance—the influence of that zone is reduced. When you go up in force you're actually reducing the relative effect. Given geometry, given system may have a larger effect at a lower force. You get a larger percentage modulation. What's interesting is that when you look at the diamond itself at the stylus tip, there are many degrees of freedom in which it can vibrate, and there may be from 20 to 50 micrograms, but their moments of inertia are considerably less. And we found that there are modes of resonance, modes of vibration, that exist in example to example in a given design, that are quite different. Because they differ in their mounting, rather than in their frequency response. So the controlled parameters are controlling something that is of no concern. They're not controlling the degree of tightness of the mounting, the exact position of the diamond, because it doesn't seem to influence these other effects. The consequence of which is that two different examples of the same pickup will wear quite differently. We find a very strong relationship between wear in both record and in diamond in the mechanical impedance at the diamond.
EDITOR: Are you aware of any pickup designers today who pursue their design efforts in the light of this information?
COTTER: No.
OTALA: Yes. Harbo Andersen at Ortofon. The MC-30 is partly the result of that kind of studies.
COTTER: He's aware of the non-Hertzian law?
OTALA: Yes, and he's also quite well aware—for instance, he also designs cutter heads—he's quite well aware of the problem of various dampings, of the damping variation of the different resonances. For instance, the Ortofon cutter head there are five major resonances, and their damping behaves differently due to the elasticity of the lacquer master when he's cutting. He describes them as highly nonlinear; he's also discussing some of the penetration effects.
COTTER: He's not published anything, though.
OTALA: No, he never publishes anything, he just designs.
EDITOR: What is the man's name again, Matti?
OTALA: Harbo Andersen.
EDITOR: Has he been there long, at Ortofon?
OTALA: Ten years at least, twelve.
EDITOR: That's interesting. The MC-30 I believe hasn't been marketed in this country yet, has it?
OTALA: I saw the first hand-made samples two months ago in Copenhagen. Two weeks ago we got the first sample into the States. The quantity is set to be 2,000 for the whole world.
EDITOR: Per year?
OTALA: No, total.
COTTER: What is the geometry of his line-contact stylus?
OTALA: Well, it's basically the same as the MC-20.
COTTER: It is not a Pramanik, even.
OTALA: No, it's not anything like that. It's extremely light. I don't know the figures, I haven't got the latest data sheets—or any data sheets. I've seen the production. The frequency response, at least in the first samples, went to about 70 kHz flat, and all the major resonances are well above the usual, the 20 kHz normal range.
COTTER: Another part of the physics of this system that's extremely important and is responsible for some of the differences between moving coils and moving magnets is that there's a considerable drag force, which is the thing that gives us the skating force. This drag force is about a third of the magnitude of the vertical tracking force and it is a variable; it varies with the signal, but not exactly with the signal, but with an even-ordered harmonic series of the signal. This is the distortion named by Rabinow and Codier in a very old paper in the early '50s "needle drag distortion." Now this force is of a very significant magnitude and is applied right to the stylus. If you constrain the system so that the stylus in effect doesn't move axially—because if it moves, you have a geometric error and the ball game is over—but if you constrain it so that the stylus doesn't move, you still have to deal with this force propagating up this "250-foot long tube" to something or other back there that's going to respond to the energy. Much the same as one can demonstrate in elementary physics lab when you have a billiard ball hooked on to a metal cylinder with another billiard ball hanging on the other end—if you rap this billard ball, the tube may not move but the force is communicated back out there to the other end. So this force which propagates back to the generator will be inclined to induce some movement in the generator that is in the axial direction. And the big difference between moving coils and moving fields is that the coil can move in two directions in which it will produce no output. Because of fringing field, there is no way that a generator system of the moving field kind can be utterly immune to axial displacement; so one is getting in effect a very interesting kind of signal out of most pickups due to this axial modulated force that's a distorted signal. It's instructive to know that when the electronic music people want to fuzz a signal, they make a full-wave rectifier circuit to do it. That's very much akin to the kind of force variation applied to the end of the stylus from this needle drag process. So on every pickup there is applied a fuzz box at the stylus. The question to be asked is, to what extent will the needle fuzz box force produce an output signal at the pickup? We've been able to simulate the sounds of some pickups by using a very low needle drag distortion pickup and injecting a full-wave rectified signal of the particular frequency characteristic of that pickup, and been able to simulate to the point where a panelist will identify the sound as the same as the sound of that pickup by injecting the needle drag component. We feel that the needle drag component is one of the strongest obvious differences between the sound of one pickup and another. Again, where they all show the same kind of frequency response, the same kind of normal IM and harmonic distortion. And this process is never measured. No one tells you what the axial force response . . .
EDITOR: This is time modulation, right?
COTTER: And ultimately this is a time modulation kind of thing, because it is a force moving the generator in and out that is essentially time modulation. Of course to this we have to add the effective vertical angle differences and the tracing distortion component, which are both time modulations.
EDITOR: Would you say this is the ultimate limitation of ordinary magnetic pickups? If this one thing were not present then theoretically they could equal moving coils?
COTTER: Yes. But it's hard to conceive of a system in which your sensitivity in the axial direction isn't going to be a serious limitation, very high. In fact, there are pickups whose axial force response is more sensitive than their lateral deflection response.
OTALA: I've been waiting for you to mention the reaction type of vibration of the record surface itself.
COTTER: Again, going back to the mythology . . .
HEGEMAN: For us old-timers, are you talking about groove resonances?
OTALA: No, no. Simply due to the fact that you have a needle which you are trying to move. That's the action; that creates a reaction. Record masses are not that big, especially considering that the record material is elastic. Although it was classified as nonsense, Jean Hiraga's paper on the table mat material influencing the sound had some truth in it.
COTTER: We find that is valid, and what is more, the most important thing seems to be the degree of contact. And what seems utterly inane are these couple of little rubber bumper suspensions for the

record which just convert it into a diaphragm so it becomes acoustically active and also mechanically active. Also realize that everything I've said about the nature of this interaction process would lead you in the direction of having a much larger vertical tracking force over a larger area and increasing the amount of energy that is being stored in that wall, and increasing this excitation. So yes is the answer. You would want to damp the record; you would want it to be in good intimate contact with a sound-absorbing medium.

OTALA: Here's a very good idea then for everybody who would like to use it. Make a vacuum cleaner type of system that sucks the record down to the surface. That's the only way really to hold it in place.

HEGEMAN: Well, how do you read instrumentation tape?

OTALA: Let's go on, because I think this is also one of the neglected factors. Due to the asymmetry of the record played, there's an interesting factor where the lateral reaction of the record being converted to vertical, so that you get a kind of cross talk. Vinyl elasticity type of cross talk from one channel to another.

COTTER: That problem exists significantly in the ultrasonic zone, where the pickup's resonance at very high frequencies, hundreds of kHz in fact, can be quite detached and very active. And twisting moment of inertia is one of the principal forms of abrasive resonance in the pickups.

OTALA: I was also talking of sheer record surface variation or vibration where lateral forces are converted to . . . or they are rotating in fact.

COTTER: What I'm saying is there's a circular kind of motion which has got at a lower energy state degree of freedom that is the vertical movement. This is very commonly the case; this is how styli twist and wiggle. Also, the magnitude of this axial force allows the stylus to do a lot of that kind of thing, skittering sort of action, so that there is applied to the signal an out-of-band ultrasonic spectrum of modulation which can by nonlinear process be demodulated and dumped back down into the audio band. And is quite influential on the apparent . . .

EDITOR: And now we're back into preamps.

COTTER: And now we're back into preamps. Because all this junk is there, and it is necessarily always going to be with a phonograph pickup preemphasized by the velocity response if you have a magnetic pickup, very involved with ultrafast phenomena; and our sensitivity to the net intermodulation of this system greatly magnified by the inherent thousand-to-one difference between the gain at lower frequencies and the gain in the ultrasonic zone. Needless to say, the time, all these effects we talked about in amplifiers, are there in spades in preamplifiers, in consequence far greater, I think, than anyone has realized.

EDITOR: I don't want to drop the subject of pickups and phono . . .

HEGEMAN: While we're on the subject, I have a couple of questions. Growing up in the phonograph industry, there were certain things we used to talk about when we had a new record. We talked about radius equalization, and we talked about the resonance of the stylus tip with the groove, or stylus and groove being a rather high-frequency rising characteristic. Now, I don't translate any of this into what we've been talking. You've been sure going through all of it, Mitch, but I don't quite bring it down to earth.

COTTER: What is interesting is that if you could get an ideal system in which you're biased at some point in this force curve, you're going to get a kind of compliance that will resonate with a singular mass point to give you a simple resonance. But the whole nature of this thing is such a nonlinear process that efforts to find the resonance are very involved with what the small-force modulations are. So that the exactitude of the resonance disappears.

HEGEMAN: So you're saying that it resonates over a *band*, instead of . . .

". . . there's a very different thing going on at the stylus . . . a lot more energy and a lot higher frequencies than anybody believed."

COTTER: Right. A lot of what people have been identifying, Stew, as stylus resonances, were resonances of the stylus system, resonances of that "250-foot long" thing in torsional and in multimodal kinds of up and down propagation effects. Many, many pickups have been built whose axial Q in the audio range is in the high numbers range. A lot of energy can bounce back and forth down that tube, whose propagation time isn't exactly trivial, if you consider what's involved.

EDITOR: Is any of that dependent on the type of pickup we're dealing with? Is a moving magnet type of pickup more prone to that particular form of resonance than a moving coil? Or vice-versa?

COTTER: No, I think that's the cantilever and system design . . .

HEGEMAN: I would think so.

COTTER: And in fact, there's another thing that merits saying, and that is that signal-to-noise ratio in the phonograph art is a very different thing from what people think it is. What I'm saying is, no one has ever played a record, actually played it completely in the sense of telling you what its signal-to-noise ratio is. In fact, there is some doubt as to what it is; it's certainly a lot better than we know, because we've been using—to carry the metaphor all the way—cassette heads to play full-track recordings all this while. Furthermore, if you talk about these resonances and this propagation of energy back and forth, we are dealing with all the sorts of things that Matti was just talking about—that is, these nonlinear time-modulating processes. And if the beam can soak up a lot of energy and kick it back and forth, its propagation time is in the microseconds to *many* microseconds back and forth with significant Q in that range. There's an ample opportunity to have all sorts of events winding up from earlier time influencing the consequence of what's going on, because the energy keeps coming back and forth and is imparted to this nonlinear stylus contact and to other nonlinear processes in the generator and beam system.

OTALA: Which, by the way, only goes to remind me of a very early edition of *Audio Handbook*, which stated very, very bluntly that despite all efforts and all kinds of metal needles, the bamboo needle still is the best needle that you can buy.

FUTTERMAN: Cactus.

OTALA: Cactus? That was bamboo . . .

HEGEMAN: Cactus or bamboo, either one.

OTALA: Yes, that's only because it's a fibrous material and also very heavily damped.

EDITOR: It's a lossy material.

OTALA: Yes, so you're getting a very, very good needle.

FUTTERMAN: I have some cactus.

EDITOR: There were some other problems there, though.

HEGEMAN: I don't know, do they still have a sharpener around?

ZAYDE: You're going to have several times the time constant that will persist two or three t as a result of inadequate termination or inappropriate termination, both at the generator and at the stylus tip.

COTTER: One could look at the cantilever as a transmission line. Then of course the differences are gross in the way in which the energy is terminated.

EDITOR: Let's zero in on some specific practical problems of audiophile or audio purist interest.

HEGEMAN: I see we just passed over my question of how about radius equalization. A very strong item back in at least old mono days.

COTTER: Radius equalization was derived from originally some observations, and then Kornei's paper in the SMPE Journal, which used the Hertzian indentor idea as a description of what this playback loss process was, and suffice it to say, that's in error. What turned out to be the case is that the short wave length effects caused changes, because the whole system was resonant within and around the audio band. And that if you could avoid that, then indeed there is an aperture process—there is a physical aperture process. The scanning stylus has a certain wave length. That aperture process is very much like the tape process—it tends to follow a sine x over x kind of relationship. Being that kind of function, as long as you stay far enough away from it, there isn't much correction. If you're into it, then you're in a bad operating condition to start with. If you're dealing with a quarter-mil kind of indentor, if you're in the line contact kind of condition particularly, then you're sufficiently outside the realm where radius equalization makes any sense whatsoever.

HEGEMAN: You're telling me that at a 6-inch radius on a 12-inch disc you get the same sound that you do on the entering and starting grooves?

COTTER: If you cut it appropriately, and if you play it back with a small enough radius of curvature, and you have little enough reactance in the stylus, the answer would be yes. You couldn't tell the difference.

EDITOR: But there is a limit to the information density you can achieve as you keep slowing down the linear speed of your groove.

COTTER: I'm saying you would want to do everything both in the design of . . . See, it becomes possible to make a radius of curvature for the aperture that is very much smaller than you might otherwise have thought possible, and still have a fairly large area of contact, if you will, or still have a fairly good signal-to-noise ratio. If you keep all your ducts in order, then with a modern kind of stylus you can play without any significant loss from the playback process. But interestingly enough, a lot of the cutters, notably the SX-68—from which, by the way, the Ortofon is quite different in this respect—the SX-74, have significant reaction effect at the cutting stylus from the smaller diameters that change what's cut onto the disc. If you correct for that, and the CD-4 people were faced with that problem, then you can make a record which inside to outside is not observably different in response characteristic. I think there's more cutting aperture loss than anyone believed. Because even if you have a 60, 50, 30-microinch radius cutter, the cutter is in effect exciting a much larger zone of influence. The hot stylus condition has more to do with easing the removal of the chip, it would seem, than it has to do with influencing the actual cutting, which isn't really a cutting, it's a tearing. You're influencing considerably more than just the surface layer of what you cut.

EDITOR: Gentlemen, where does all this leave us with respect to desirable pickup design and desirable preamplifier design? Which is really I think what our subscribers would like to know.

COTTER: My purpose in talking about it was to show that there's a very different thing going on at the stylus.

EDITOR: We are very glad that you talked about it.

COTTER: It's relatively new information. It's a very different thing going on at the stylus. The consequence of it is threefold, of serious concern. There's a lot more energy and a lot higher frequencies than anybody believed. This energy causes reactions in the pickup that hadn't been suspected, although Rabinow and Codier and others pointed to the existence of the potential 25 years ago. That the pickup, being a velocity sensor, presents to the preamplifier, in consequence of this—particularly if you have, say, a moving coil structure that's not imbued with these low-pass filter properties—a very much more difficult signal to handle in face of the inherent equalization that you're applying than has been considered. For instance, in a power amplifier we may have high-frequency effects, time modulation processes, things of this sort, that will produce results in a flat amplifier that we have judged to be serious problems. Now what we're doing is, we're suppressing our awareness of it by in effect removing our sensitivity to it by a factor of 100 to 1000, and then boosting the response to the lower frequency consequences of its existence by a factor of 100 to 1000. So in effect the equalization demand and the preamp problem are vastly more serious than has been considered.

OTALA: There's a number of problems in that respect. The first one being, I believe, that we're exciting quite a lot of frequencies at the pickup mechanical resonances. But also at the preamplifier, which is normally a feedback type of system having a normal phase margin and a peaking response somewhere. They behave the same way; they're complex conjugate poles. Since they are complex conjugate poles, any type of transient excitation simply gives us more or less a burst of "carrier" frequency there. It's ringing up there; it's creating a carrier type of signal; and this carrier type of signal occurs at frequencies where both the amlifier and the pickup are highly nonlinear. That mixes with all the ultrasonic signals present at that frequency region, and that's what you mentioned.

COTTER: Plus the base band, which is a Hilbert transform, which is then going to translate with incoherent distortion products back down into the audio band.

OTALA: Yes, it's going back to the audio range. This is one of the problems. The second problem is that most of the preamplifiers as far as RIAA correction is concerned, do not have a specified frequency response, say above 30 kHz or whatever.

COTTER: The missing pole.

OTALA: Yes. Therefore we have had some dramatic difficulties in the TIM psychoacoustic experiments, for instance. We were forced to go to the STM-72 transformer with the Ortofon MC-20 pickup. And that was simply because of the fact that it filtered our ultrasonic response. There's a lot of this kind of effects, and although it is not strictly what we are discussing now, it might be good to tell you how we did select the records and all the components. It was very simple in fact. We had the distortion generator there, which we could . . .

COTTER: This is a digital synthesizer.

OTALA: A digital synthesizer. And we could adjust the additional distortion that we created into the signal. We had a measurement system which showed us the percentage of distortion being generated. And the simple trick was—the only one that worked, in fact—was to replace or modify each part of the system so that the threshold where the added distortion just became audible was at the lowest. By this way, by adjusting the stylus pressures, changing pickups, changing preamps or transformers, and doing this and that we finally came to a very, very sensitive system. The same system applied for selecting the records. And that was simply this—you play a record, you note what is the distortion level which just now becomes audible, take the next record and take again the percentage level—if it was lower, then this record has less masking. And that's the kind of effect we have to play here too.

COTTER: Let me tell you of an interesting consequence of this understanding that was revealed to us in some experiments we did with respect to vertical angle. And that's an observation that's continuing, and I think others have now formed similar impressions. It's always troubled me that a shockingly small variation in the vertical angle, usually in the positive or increasing vertical angle direction, would cause very marked changes in the sound character. Usually a kind of hardness and brightness, if you were to put metaphors on it.

EDITOR: To reinforce that, one way of tuning the VTA by ear is to get to that point where the sound hardens up, and then back off a little bit. That has been the point of greatest clarity.

COTTER: Now there are two things about that condition that have always troubled me. One is that when I measure the actual frequency modulation produced by a pickup, I would never come up with that angle, usually with a much higher angle, as the effective vertical angle of the pickup. In other words, the tendency by ear was to come out with an angle that was lower. Further, nothing in my repertoire of understanding could quite suggest a mechanism that was so abruptly changing in the region of this effect, because further increase did not cause any overwhelming collapse. We now think we understand that process as a trigger process; that in effect when one varies the vertical angle, what you are doing is varying the *slope* of the response of the stylus to the wave form in such a way as to modulate its slope, because changes in vertical angle are in effect changes in the speed with which you're reproducing.

EDITOR: The slope of which response?

COTTER: The slope of the needle's motion. Its time derivative is being changed. Its rate of change is being changed very significantly by changes in vertical angle. In effect, what you're doing is getting a phenomenon within the electronics of the system wherein you are getting into some kind of marginal, incremental, slew-type phenomenon that is a trigger-like phenomenon.

OTALA: It is not really slewing, but it is that kind of change which seems to . . .

COTTER: It's that kind of change, for want of a better word. Anyway, we think we understand this, because if you dissect the topology of most of the existing systems that include equalization in the feedback loop—and it's important to notice that irrespective of which particular combination of artifice and technique you encounter, that the topology of most of these systems is essentially that kind of topology. That it resembles very strongly the basic Schmitt trigger circuit. You have a scheme within which, if you remove the feedback, and consider that the error signal can expand rapidly and dramatically, you are getting into an energy storage situation that on a short-term basis resembles a trigger snap. In effect, the character of the sound and the kind of marked influence for these marginal levels, affects it. There's one other experiment that you can perform to, inferentially at least, see whether there is something like this taking place. Since what we're talking about is a velocity-responsive sensor, and when you change the vertical angle you are in effect changing the amplitude of the pulse as well as its rate of rise, one should be in a position to suppress this critical moment, this critical angle, by simply reducing the magnitude of the signal. Not afterwards, but as presented to the preamp. A moving coil pickup makes it very easy to do this; all you have to do is apply a loading resistor to trim the level. When we tried this, remarkably, you could reduce the disturbance, you could go below the threshold, in effect. You didn't get the hardening. You could come up a little more before you got the hardening. We've been exploring this for a couple of years. I haven't been sure just how much of each contribution there was, and I've become convinced recently that this is an overwhelming effect in most systems. In fact, when you get a system that is incapable of triggering because you haven't got that topology, a feedbackless passive equalizer system, then you don't find any of this effect, and the null and the vertical angle correspond with the measured effect, and in fact it's much less critical than you otherwise find.

EDITOR: We found in our equipment evaluations here that with the preamps we've been leaning toward most recently, the VTA sensitivity is considerably reduced. It has become less critical. It was supercritical with less good preamplifiers. On the other hand, we haven't found that it's necessarily restricted to preamps that are equalized in the feedback loop. As a matter of fact I don't think there's anybody sitting around this table here today who equalizes in the feedback loop.

HEGEMAN: Strangely enough.

COTTER: Maybe, but there still are probably TIM-like time modulating processes which this could improve.

OTALA: Except in all the preamps tried by Alvin Foster. He just published a paper and said he didn't find any.

COTTER: Any what?

OTALA: Any TIM in any kind of preamplifier he tested, starting from—well, he had 50 or so.

EDITOR: Who was this, Matti?

OTALA: Alvin Foster, the Boston Audio Society founder.

RAPPAPORT: Well, they're the ones who called you a charlatan anyway, right?

OTALA: I would like to come back to my statement in the beginning, that I believe strongly that it is important to reproduce the amplitude faithfully and the first derivative faithfully, and the second derivative faithfully—we just proved that. Let's put it this way: we have a series of transfer characteristics. The first transfer characteristic is the output voltage versus input voltage. This should be a straight line. The second one is the dynamic transfer characteristic, and that is output rate of change versus input rate of change, which should also be a straight line. The third is the second derivative —how would you like to name that, that's another thing.

COTTER: The rate of change, acceleration . . . The second moment, the third moment, the fourth moment . . .

OTALA: So all those plots should be linear and should be accurately reproduced. That is probably the consequence.

COTTER: Well all of these come down to non-time-dispersiveness, if you consider that rate puts its heavy emphasis on the time at which it occurs.

OTALA: If you take the nth derivative, and plot that nth derivative of the output signal versus the nth derivative of the input signal, all derivatives with respect to time, and that nth plot is straight and linear, you don't have any time dispersion. That's the conclusion.

COTTER: It's important to go back again to the purely geometric description of what goes on in the phonographic system, and realize that with a rounded contact on a stylus compared to the sharper contact of the cutting stylus, that there is a time modulation that occurs such that as you go uphill, the point of contact moves in front, as you go downhill the point of contact moves around behind you. So one has a time-swinging effect, in which the rate of swing is proportional to the second moment, the second derivative; this acceleration component is modulating the time position, and that that is the tracing process. Its very nature is one of time modulation. That the vertical angle errors are a displacement-proportional time modulation, and that lateral tracking error is the same kind of thing, and that even some of these elastic effects which store energy can return essentially a time-modulated displacement force. And in fact, all of the phonographic disturbances have an intrinsic characteristic of being time-modulated.

EDITOR: And not all synchronous, right?

COTTER: And not all synchronous; obviously displacement in one case and acceleration in the other case, and an energy storage thing which is a complex thing because it varies with frequency, if we may reintroduce that specter.

OTALA: It also varies with the place on the groove where you are because you have different geometry, whether you're in an outer or inner groove, or outer or inner curvature.

COTTER: Because it's wave-length dependent as well.

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 . . .

To be concluded in the next issue.

Speaker Systems, Large and Small: Updates and New Developments

By the Staff of
The Audio Critic

Other than a new medium-priced speaker of Reference B quality, plus some improvements in existing good designs, we haven't found much this time to get excited about—but maybe that's enough for one go-around.

If you've read the preambles to the speaker surveys in our last three issues, you know pretty much where we stand on the subject of speaker design and evaluation. We have no fresh insights or important new criteria to add at this point, which is probably a good thing considering the unprecedented amount of theoretical material presented elsewhere in this issue (seminar transcript, stylus article, etc.). At the risk of underestimating our subscribers' attention span, we'll skip the orientation lecture this time and proceed directly to the reviews.

Beveridge 'System 2SW-1'
(follow-up)

Harold Beveridge, Inc., 505 East Montecito Street, PO Box 40256, Santa Barbara, CA 93103. Beveridge Cylindrical Sound System, Model 2SW-1, $7000 the pair (including plug-in direct-drive tube amplifiers, HD subwoofers, solid-state bass amplifiers, electronic crossovers and CM-1 control module). Unlimited warranty on all parts except tubes (one year); five-year warranty on all labor, including sonic updates. Tested system supplied by manufacturer.

The 2SW-1 is the modified Beveridge system we summarized in our reference issue (Vol. 1, No. 6) without actually having put it through all our tests. Now that we've gone over it we can report that it's essentially the same system as the 2SW as far as sonic performance is concerned, with minor improvements. In other words, it's still the best speaker system from a single manufacturer in our opinion, though not necessarily the best speaker system that can be put together as a hybrid by the technically adept audio enthusiast.

The dynamic headroom of the 2SW-1 still isn't exactly what the doctor ordered, although subjectively it seems ever-so-slightly better than that of the 2SW, which we didn't have side by side for comparison. The extreme highs are still slightly rolled off; the bass, however, has been definitely improved. The woofer Q is now 0.7, as claimed, and stays there with increasing drive; the -3 dB point is 36 Hz; THD is quite low at all bass frequencies, even at fairly high levels. We still can't see, though, why the woofer of a $7000 system shouldn't be flat down to 30 Hz or so, especially since high efficiency isn't a requirement for a good match to the electrostatic main speaker.

As you may have heard, Beveridge is about to come out with a new speaker system utilizing basically the same 6-foot electrostatic transducer and acoustic lens, but with conventional transformer coupling to any power amplifier supplied by the consumer and with two woofers in each of the 6-foot enclosures instead of separate bass commodes. It's our impression that this might turn out to be a more practical and convenient system with perhaps even superior sound. The price will be approximately half that of the 2SW-1.

DCM 'Time Window' Pedestals

R. S. Park Audio Associates, 5 Sunrise Plaza, Valley Stream, NY 11581. DCM Time Window Pedestals, $65 the pair. Tested samples on loan from distributor.

As our orginal review stated, the DCM Time Window will sound even better when raised off the floor. It's partly the peculiar tuning of the vented bass enclosure, partly the height of the drivers with respect to the listener's ears.

These new pedestals do the job neatly and painlessly. Made of wrought iron and shaped to cradle the curved bottom panel of the Time Window, the pedestals securely elevate the speakers 9 inches above the floor, with audible benefits in airiness and reduced bass whomp.

It must be added that a revised edition of the Time Window has just come out, which we haven't tested yet. It has entirely new bass/midrange drivers, and the result may be a whole new ball game. A review is scheduled for the next issue.

Dennesen Model 180 and SW-II

Dennesen Electrostatics Inc., PO Box 51, Beverly, MA 01915. Model 180 electrostatic/dynamic compact speaker system, $440 the pair. Model SW-II subwoofer to match, $275 each ($550 the pair). Five-year warranty on drivers and crossovers; two-year warranty on power supply and energizers; customer pays all freight. Tested #2194 and #2195 plus unmarked woofers, on loan from manufacturer.

This is the speaker debut of a young company that appears to us very serious and sincere about good sound. Their avowed intention is to provide satisfaction to highly discriminating audiophiles at dramatically low price. A brave idea, not quite brought to fruition in these speakers.

The basic building block of the Model 180, which is a box speaker only a little over one cubic foot in internal volume, is a round electrostatic tweeter element with a 2½" diaphragm. By itself, this is a neat little unit; we found it to be quite flat and smooth in nearfield response all the way up to 43 kHz, with little or no ringing. The trouble is that Dennesen uses a cluster of five of these tweeters, deployed in a more or less hemispherical formation, in an attempt "to achieve a coherent hemispherical wave front resulting in uncanny spatial placement with precise imaging, almost anywhere in your listening room" (it says in the blurb). The concept is shaky both mathematically and in actual practice; you can't synthesize a hemispherical wave front by jamming together five circular clamped radiators with edge effects. There are bound to be some pretty grim interference patterns, and indeed our measurements revealed them in profusion. The semi-nearfield frequency response of the cluster is of the ± 7 dB snaggletoothed variety as a result, and tone bursts display severe cancellation and reinforcement effects. Add to that the fact that the 8" woofer, which is crossed over at 1250 Hz, is out of phase with the tweeters, and you begin to see that coherence is hardly the word that applies here. Pulses are, of course, irreproducible with the tweeters pushing when the woofer is pulling, especially since the reversal of polarity is plunk in the middle of the passband.

Speaking of that 8" sealed-box woofer of the Model 180, it shows a 4½ dB peak at approximately 50 Hz and is rather poorly damped. The Q appears to be in the neighborhood of 1.75, rising to 2.0 and beyond with increased drive. Sloppy.

In our listening tests we found the speaker to be hard, bright, sizzly and fatiguing. Since the interference patterns and peakiness are worst right where tweeter cluster first cuts in, from 1.25 kHz to 3 kHz or so, this irritating quality doesn't surprise us; the ear is extremely sensitive in that range. The manufacturer insists that placement of the Model 180 in the room is extremely critical, and that of course is necessarily true wherever reinforcement and cancellation effects dominate; it's like insisting that body placement is extremely critical on a lumpy mattress. How about eliminating the lumps instead?

As for the Model SW-II subwoofer, we don't quite see the point in it. It's claimed to be designed specifically to match the Model 180, yet its frequency response doesn't extend appreciably lower, nor is it significantly better damped. We measured ± 2 dB small-signal response down to 48 Hz and a Q that varied from 0.7 (excellent) to 1.5 (underdamped), depending on the amount of drive. The frequency response was also affected to some degree by the drive, indicating a voice coil that comes out of the gap.

Overall, we're unimpressed by the Dennesen speakers, even though we appreciate this company's recognition of the inherent superiority of the electrostatic principle and of the fact that the high cost of electrostatic speakers isn't a law of nature but a marketing phenomenon that can be altered.

Fried Model C

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 #C1047K and #C1048K, on loan from manufacturer.

This is the top part of Fried's new $4000-plus Super Monitor, which we haven't tested. As a separate satellite, the Model C is very similar to the B/2 reviewed in the last issue; it has exactly the same dome tweeter, a considerably heftier 6" bass/midrange driver, and a heavier enclosure that tapers toward the top. It should by all rights sound a little better (take a look at that price!) but it happens to sound considerably worse. We found it very strident and fatiguing, especially nasal and cutting on solo strings, but almost equally unpleasant on most instrumental combinations as well as voices. This assessment proved to be quite shocking when told to a number of people who were familiar with the speaker, so we must contemplate the outside possibility that something was wrong with our samples. If there was, it was the same defect in both speakers, which would be quite a coincidence.

We believe that the stridency is due to a broad peak in frequency response centering on 3.8 kHz, where the ear is extremely sensitive. This is one case where the time domain doesn't appear to be the source of the problem, as we found pulse replication quite accurate to 0.2 msec (only the very best speakers make it to 0.1 msec) and saw little or no ringing on tone bursts.

Aside from the somewhat alpine frequency response profile, the greatest peculiarity of the Model C is the pair of ½-inch holes drilled into the otherwise tightly sealed enclosure. These provide an ineffectual sort of resistive loading that makes the enclosure conform neither to optimized vented-system parameters nor to a pure sealed-box model. It's the worst of both worlds. A.N. Thiele, the man who practically invented the mathematical approach to woofer alignment, has an admonition against this very technique in his classic paper. The result is a mistuned box with bollixed-up damping characteristics; the response to a step function resembles that of a system with a Q of approximately 2.0, but with increased drive the volume velocity through those constricted vents becomes enormous at 100 Hz and below, so that the holes actually whistle and no accurate microphone measurement of any kind can be made. It's quite clear that the woofer Q itself is much too high and those little holes in that little box can only make matters worse instead of better. The fundamental resonance of the system appears to be at approximately 100 Hz, where there's a nice fat hump just as you'd expect. The rationale for all this as presented by Fried Products in their literature is unrelated to any system of physics or mathematics known to us.

Perhaps we're making too much of the poorly controlled bass of the Model C, since it's sonic impact is dominated by the stridency in the treble. In any event, we suggest that its designers listen carefully to the cheaper and less efficient Model B/2 again as a fair example of neutral and uncolored speaker sound.

Fried Model W

Fried Products Co., 7616 City Line Avenue, Philadelphia, PA 19151. Model W three-way speaker system, $640 the pair. Two-year warranty. Tested #2008 and #2009, on loan from manufacturer.

Word had been passed to us just before we received the Model W that we were going to prefer it to the DCM Time Window and probably nominate it as our Reference B speaker. That turned out to be a rather unsuccessful bit of prophecy but we're mentioning it because it indicates where the manufacturer positions this new 3-way, 2-cubic-foot model.

To us the speaker sounded grotesquely colored, altering some instruments almost beyond recognition both in size and timbre, and giving voices a thick, funky quality, as if the singers and talkers had colds. Our listening tests didn't last very long.

This time the trouble is definitely not in the frequency domain; we measured extremely flat response from 50 Hz all the way up to 20 kHz, with just a very slight and smooth rise at 15 kHz. The problem is in the time domain, beginning with the fact that the 4" midrange driver is out of phase with the 8" woofer and the 1" dome tweeter. This throws two octaves in the middle out of sync with the rest of the spectrum, resulting in severe additive and subtractive interference patterns on tone bursts and badly impaired pulse replication. This sort of thing is invariably audible and can't be rationalized away.

As in several other Fried systems, there are also bass anomalies. The quasi-third-order vented system appeared to be horrendously underdamped as the speaker was first delivered to us; additional strips of acoustical foam supplied by the manufacturer changed the picture somewhat. Without the latter, there was an 8 dB hump at 70 Hz (the only interruption of the smooth overall response profile), and a high-amplitude step function caused 80 msec of ringing. Ouch. Harmonic distortion increased dramatically from 70 Hz downward. With the extra strips stuffed into the vent, the amplitude of the hump was reduced and the bass response looked more like that of a somewhat underdamped sealed box, extending down to about 40 Hz with a Q of approximately 1.5, wandering to higher values with increasing drive. Better, but still far from optimum alignment.

You can stuff a turkey but you can't change it into a bird of paradise.

Pyramid Model T-1
(Improved)

Pyramid Loudspeaker Corporation, 131-15 Fowler Avenue, Flushing, NY 11355. Model T-1 Ribbon Tweeter, $1175. Three-year warranty. Tested factory-modified samples, owned by The Audio Critic.

The only serious fault of what we called the world's best tweeter in the last issue has been corrected, at least to some degree. The 5-position filter/attenuator has been redesigned, so that it comes much closer to allowing the same frequency response profile in all positions. We still discern something of an energy-storing, Q-ey hump just past the "corner" of the high-pass filter in the 0 and -2 dB positions, but not as elevated as before. In the -4 and -6 dB positions, most likely to match the efficiencies of typical systems, the hump collapses quite satisfactorily, and in the -8 dB position, which will match only very inefficient systems, the response of the tweeter is ruler-flat from 2.5 kHz to 30 kHz. With a separate level-controlled amplifier channel driving the tweeter, it's the -8 dB position we recommend. Tone bursts elicited no ringing at any frequency in the modified T-1.

Actually, by playing around with various networks, it's possible to equalize the response of the tweeter within ±2 dB to 43 kHz; with the new factory version we don't really think it's worth the bother. We've also determined that the inherent rise time of the ribbon is of the order of 8 microseconds, which is faster than the human ear. Very few, if any, other tweeters can make that statement; maybe that's why they create the subjective impression of an "aperture loss" next to the Pyramid T-1.

For the upper two and a half octaves of the audio spectrum, this remains our standard of excellence.

Symdex 'Sigma'
(Improved)

Symdex Corporation, 12 Irving Street, Framingham, MA 01701. 'Sigma' loudspeaker, $598 the pair. Tested samples on loan from manufacturer.

This highly inefficient little two-way speaker system has a new woofer but hasn't changed a great deal in sound quality—that's about the extent of the news. For the full discussion of the design see our last issue; here we'll discuss only the changes.

The new woofer in the same sealed box appears to result in a higher Q; we measured 1.1 to 1.2 on a dynamic basis, without noticing any tendency this time to migrate to higher values with increasing drive. The slight reduction in damping results in a few more cycles of bottom range; the -3 dB point is now 56 Hz. The overall frequency response of the system is very smooth all the way up to 20 kHz; the "sweet spot" for the best reading has moved closer to the woofer end, it seems. Pulse response is very decent but, interestingly enough, not nearly as excellent as before; tone bursts elicit virtually no ringing or extra cycles anywhere.

The speaker still sounds rather lifeless, constricted and uncomfortable on dynamic program material in a large room, perhaps somewhat less obviously so than in the previous version. We still couldn't live with it as our prime source of music. On the other hand, we must admit and emphasize again that at low levels no other moving-coil speaker system known to us sounds as uncolored and "electrostatic." Audio purists in dormitory rooms and other small spaces might find it just perfect for their needs, as long as they use a very good and powerful amplifier to drive it.

Thiel Model 03
(follow-up)

Thiel Audio Products Co., 4158 Georgetown Road, Lexington, KY 40505. Model 03 floor-standing coherent-source loudspeaker, $775 the pair. Tested #0119 and #0120, with equalizer #0059, on loan from manufacturer.

The production Model 03 with the revised midrange arrived only a few weeks after our publication of the "interim report" in the last issue; with slightly better timing this review could have been the original one. Sorry about that. The delay allowed one interesting misconception about the speaker to come out in the open, however; some of our readers are apparently under the impression that Jim Thiel, the personable young man who designed the Model 03, is none other than A. Neville Thiele, the great Australian investigator of the mathematical analogies between high-pass filters and vented loudspeakers. The fact is that Neville is old enough to be Jim's father and spells his last name differently; the vented bass enclosure of the Model 03 was aligned not by Thiele but by Thiel according to Thiele—and not quite accurately, as we shall see.

The midrange of the final production model is indeed greatly improved; in fact, the overall sonic impression made by the Model 03 is quite favorable, especially on first listening. Prolonged exposure to it made us conclude, though, that what at first appeared like excellent clarity and resolution was actually a bit of highlighting zippiness, creating an italicizing effect without revealing ultimate inner detail. When we put our old standby, the DCM Time Window, next to the Thiel, it became immediately apparent that the latter was less transparent and balanced in sound, adding some spurious information to the signal at all times.

As for the electronically equalized vented-box 10" woofer, there seem to be some problems. Sixth-order Butterworth alignment is claimed, but the Q looks a bit high to us for that to be true; there's a 3 dB elevation centering on 70 Hz and spanning 60 to 100 Hz. The response goes way down, though; the -3 dB frequency is 20 Hz and, since the corner is very sharp with a sixth-order slope, 21 Hz is already up on the 0 dB line. Before you say "Wow!" we must quickly point out that this is in no way like 20 Hz response out of a 50-cubic-foot box with four 15" woofers; it's strictly a small-signal tuning characteristic, and the distortion is very high between 20 and 30 Hz. The lowest tone that looked really clean to us was 38 Hz. When somebody steps on an organ pedal way down there, the result is mostly hash. It would have made more sense to tune that 10" woofer and small box to a higher frequency.

The overall frequency response of the Model 03 is quite flat up to 15 kHz, except for a fairly broad dip centering on approximately 900 Hz. The -3 dB point on the top end is at 20 kHz. Pulse response looks good, as it should in a speaker claimed to be a "coherent source," but the solid angle over which pulse shapes are accurately retained is extremely small, so that the coherence is mostly academic. Tone bursts revealed severe interference patterns caused by the lack of physical separation between the dome tweeter and the midrange driver. At the "sweet spot" everything is fine and dandy, but move the measuring microphone just a hair and there are cancellations and reinforcements all over the place, just as if the drivers were ringing. At 3.6 kHz the midrange *is* ringing, for real. These effects may very well account for the zippiness we heard.

It also has to be added that the electronic equalizer supplied by Thiel, which must be inserted between the preamp and power amp or into the tape monitor loop for correct woofer response, isn't necessarily an audio component of absolute transparency. That's a minor point, however. We still believe that the Model 03 should be rated as a good speaker system, better than most, but certainly not one of the best.

Vandersteen Model II

Vandersteen Audio, 1018 South Mooney Boulevard, Visalia, CA 93277. Model II floor-standing 3-way speaker system, $860 the pair ($880 east of Denver). Matching 6" high metal stands, $50 the pair. Tested

#2638 and #2639, on loan from manufacturer.

Let's start with a paradox. The Vandersteen Model II is the best-sounding speaker system known to us anywhere near its price and is therefore our new Reference B selection. At the same time we very much disapprove of several aspects of its design—more vehemently so than we might in the case of some other speaker we like less.

Now let's qualify all that. To our ears, the Vandersteen sounds more accurate—meaning smoother, more neutral, more finely detailed—than the DCM Time Window, our previous top choice in this category. On the other hand, we haven't tested the very latest Time Window, which has a new set of bass/midrange drivers and is claimed by DCM to be considerably improved. As for the design flaws of the Vandersteen, without which it would be a truly superior speaker, there are two that bother us in particular. One is that the 2" midrange dome is out of phase with the 1" tweeter dome and the 8" woofer. The other is that the 12" passive radiator in the fourth-order vented bass enclosure doesn't track correctly with the woofer; the system is mistuned. (The active woofer has its null at 42 Hz whereas the passive radiator peaks at 50 Hz—a 19% error.) You can hear both of these defects—the first as a marginal smeariness or tonal confusion, the second as a bit of "woofing up"—but the overall sound of the speaker is still so good that we can't demote it from the head of its class.

Why should that be so? For one thing, the Vandersteen deals very successfully with the problems of diffraction and secondary radiation, by means of an ingenious free-standing, unbaffled mounting of the upper two drivers. These drivers are, in addition, of high quality, resulting in smooth and reasonably flat response all the way up to 30 kHz, without obvious ringing at any frequency. Below 110 Hz the bass is a little lumpy, however, for the reason already mentioned. We'd say that the nominal bottom of the speaker's range is somewhere in the upper 40's. Pulse response is, of course, hopelessly messed up by the out-of-phase midrange. Luckily, the latter covers almost four octaves, so that it tends to dominate the perceived tonality instead of displacing a narrow piece out of the middle like some other out-of-phase midranges. That might well be the major secret of the speaker—plus the fact that its first-order (6 dB per octave) crossover slopes can't possibly get it into time-domain trouble.

Stereo imaging is one of the strongest points of the Vandersteen, probably owing to the nondiffractive deployment of the drivers. The image is stable and it's located in back of the speakers as it should be. We must caution you, however, about the extremely broad range of the midrange and tweeter controls. Without a calibrated microphone it isn't easy to find the positions that result in the flattest response. It's a slightly quirky speaker, any way you look at it.

But, unlike so many upper-medium-priced speakers, it does sound like music. Take that from a critic who actually has a deep suspicion of "musicality" as the favorite cop-out of those who lack objective criteria.

XRT 50

(interim report)

The XRT Group, PO Box 8, Route 4, Menomonie, WI 54751. Model 50 floor-standing 3-way speaker system, $938 the pair. Tested unnumbered samples, on loan from manufacturer.

At press time we're informed that this small speaker manufacturer is in the process of merging with another company, so that their "future marketing and manufacturing plans are in a state of flux." Since this particular speaker may therefore still surface somewhere, altered or unaltered, as a new product, it will be useful to summarize its characteristics without going into details.

The vented bass enclosure is claimed to be Thiele-aligned but looks grossly misaligned to us. An active equalization module was originally planned to be a future add-on, which may explain the discrepancy. The midrange is out of phase with the woofer and tweeter (here we go again!) and has rather ragged response. The tweeter is excellent, with on-axis response to 35 kHz and very good dispersion up to 12 kHz. The sound is a bit on the zippy and aggressive side, probably on account of large peaks, valleys, suckouts and interference patterns in the midrange and lower treble. Efficiency and headroom are relatively high; overall construction appears to be of solid quality.

A follow-up review will be published should the XRT 50 become a widely marketed product.

Recommendations

There's only one change here since the last issue but it's an important one, from the DCM Time Window to the Vandersteen Model II. Note, however, that an allegedly improved version of the Time Window will be reviewed in the next issue.

Best speaker system: Reference A of *The Audio Critic* (see article on reference systems).

Best speaker system from a single manufacturer: Beveridge System 2SW-1.

Best speaker system per dollar: Vandersteen Model II.

* * *

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.

Symmetry ACS-1

John Curl's Perfectly Coherent Electronic Crossover

Symmetry Audiophile Systems, 101 Townsend Street, San Francisco, CA 94107. ACS-1 Active Crossover, $650. Tested #0573, on loan from manufacturer.

We have two misgivings about reviewing any separately available, all-purpose electronic crossover like the Symmetry. One is that there's no such thing as an all-purpose filter slope, whether it's 6, 12 or 18 dB per octave. The correct choice of slope depends both on the passband and on the out-of-band characteristics of each driver used, as well as on the total speaker system design concept. For example, the 12 dB per octave low-pass slope of the Symmetry would be much too gradual for the Janis W-1 woofer, which needs to have some rather ferocious out-of-band peaks attenuated as sharply as possible. The other reviewing problem is that the sonic flaws, if any, of the crossover are virtually inseparable from those of the speaker system whose signal traffic it directs—are the motorists or the corner policeman responsible for the traffic snarl? A simple bypass or substitution test is just about impossible in the case of an active crossover; we have no "straight-wire" filter networks for comparison, nor even a "reference" electronic crossover.

Those reservations out of the way, we can report that the Symmetry ACS-1 is a truly excellent unit, adding little or no subjectively perceivable coloration to the remembered sound of various electronic chains into which we've inserted it. Perhaps an occasional touch of barely evident hardness or brightness, perhaps not even that; as we've said, these aren't genuine A-B tests. And on the best bench the ACS-1 is nothing short of amazing. You can actually feed a square wave into the input, sum the signals from the low-pass and high-pass outputs, and get a perfect square wave back. We were able to do this regardless of the selected crossover frequency, which is continuously variable from 45 Hz to 4.5 kHz.

John Curl, who designed the ACS-1 (as well as the Mark Levinson JC-1 and JC-2, back in the days when he worked for Mark), believes that such waveform coherence is every bit as essential in a crossover circuit as in any other section of the audio signal path, whether upstream or downstream. We wholeheartedly agree; indeed, we can't see how anyone can disagree (except certain Bostonians). Interestingly enough, no other separate electronic crossover known to us exhibits this kind of total coherence, not even the vastly more expensive Mark Levinson LNC-2. As a consequence, you'll be way ahead with the ACS-1 in any attempt to achieve even a semblance of coherent alignment in a biamped or triamped speaker system.

On the other hand, since the ACS-1 is most likely to be used for biamping systems with separate subwoofers, it should be pointed out that coherence becomes less and less audible as you go lower in crossover frequency. In our experience, it makes absolutely no difference at 100 Hz, for example. From a few hundred Hz on up, however, and especially in the kHz range between midrange and tweeter, it becomes quite significant and shouldn't be neglected.

Our overall recommendation, then, is this: If you feel sufficiently qualified to "roll your own" biamped or triamped speaker system (are you sure?) and if you've determined that you don't need very steep crossover slopes (such as 18, 24 or possibly 36 dB octave), we don't see how you can go wrong with either one or two Symmetry ACS-1 units. It's a well-engineered piece of equipment that will perform exactly as its makers claim.

The Two Most Interesting Power Amplifiers for the Audio Purist

By the Staff of
The Audio Critic

One of them gives you the cleanest, most accurate, most natural sound available so far, regardless of all other considerations. The other sounds almost as good, costs less than one fourth as much (22 cents on the dollar), and is thoroughly practical to boot.

If there's one subject that doesn't need further coverage here after our State of the Art seminar, it's power amplifier circuit design. In the aptly chosen words of Stew Hegeman, we beat it to death. If you want to know what theoretical design criteria are considered important by some of the best engineering minds in the business, please turn forthwith to the small-print transcript in this issue. We certainly won't add another word to theirs, except to remind you that our laboratory test methods and listening setup were explained in some detail in the last three issues, with rationales and updates as we went along. Copies are still available to new subscribers.

Let's proceed then directly to the reviews.

Hafler DH-200

The David Hafler Company, 5817 Roosevelt Avenue, Pennsauken, NJ 08109. Model DH-200 stereo power amplifier, $399.95 wired. (In kit form, $299.95.) One-year warranty; manufacturer pays return freight. Tested sample on loan from manufacturer.

Our killer reputation notwithstanding, we can think of no greater editorial delight than to be able to recommend a truly superior piece of audio equipment at a reasonable price. The Hafler DH-200, either wired or as a kit, fills that bill about as neatly as anything we've tested so far. It's not only the best power amplifier *value* known to us; it's also one of the three or four best-sounding power amplifiers regardless of price. Now anyone with $400 can afford to be a finicky ultrapurist when it comes to power amps; if he's willing to solder two already assembled and wired channels to the power supply and do a few little screwdriver chores, he can swing it for $300. (According to Dave Hafler, that's not cheap; it's all the others that are overpriced.)

In our listening tests, the $1800 Rappaport AMP-1 was the only power amplifier we found unequivocally superior to the DH-200, and not by a wide margin. Both were considered by our various auditioners to be amazingly transparent, uncolored, sweet, solid, and focused in sound, but the DH-200 seemed to have just the slightest trace of "shimmer" on top by comparison—and only by comparison. Also, the Rappaport has a huge dual power supply, so it hangs in there on current-draining low-impedance loads a bit more consistently, with occasionally audible advantage. Switching to the Hafler after long exposure to the Rappaport is by no means a serious letdown, though; the minor retrogression in sonic purity is something we feel we could live with if we had to.

Against all other comers, the Hafler stood up brilliantly. The latest Futterman H-3aa, when happily feeding relatively high-impedance loads, was found marginally preferable on some program material, but the Hafler was preferred just as often if not more so; the two can certainly be considered sonically comparable. The excellent Audionics CC-2, our previous best-buy choice, sounded distinctly more "electronic" and constricted than the Hafler, although again only by direct comparison. The Mark Levinson ML-2 wasn't available, but our previous ranking of early versions of it as a hair below the Futterman should suffice as a guideline. All other power amps were hastily retired after a few minutes of comparison with the Hafler.

This kind of comfortable superiority is seldom due to good clean engineering alone; one begins to suspect some kind of very fundamental technological edge on the competition. That may well be the case; the DH-200 is the first American amplifier to utilize Hitachi's new power MOS FET's. What's more, Erno Borbely, the Hafler engineer responsible for the design of the DH-200, created a very different circuit to go with the power MOS FET's than what you'll find in the few Japanese amplifiers that took early advantage of this new output device. It seems he was able to use quite a bit of feedback without ill effect, since the MOS FET's introduce very little front-to-back delay (see also the seminar transcript on this subject). With 2 microfarads shunting the 8-ohm load resistor, we measured 30% overshoot on square waves, damped within less than 20 microseconds. With 1 microfarad across 8 ohms, the overshoot dropped to 15%. Into a purely resistive load, there was no overshoot. Not that any of this proves a whole hell of a lot; it does indicate, though, that there's plenty of feedback in the circuit and that it's used in an intelligent, controlled manner. Whether that tiny residual shimmer on top is feedback-related is a moot point. (Again, we refer you to the seminar transcript for the more sophisticated aspects of the matter.)

Our routine program of measurements revealed absolutely no vices in the DH-200; THD was especially low at all frequencies and all power levels. We found the official power rating of 100/100 watts into 8 ohms and 150/150 watts into 4 ohms to be extremely conservative; you can definitely count on a bit more. The power supply (single transformer, two hefty electrolytics) becomes the ultimate limitation on current capability; with 4-ohm loads the purist may want to consider using two DH-200's with only one channel connected on each. That's still only $800 for a super stereo amplifier. We're also told that a $25 bridging device for converting the DH-200 into a 300-watt mono amplifier (at 8 ohms) will be available shortly. That should put the fear of God into makers of multikilobuck exotic amplifiers.

As far as reliability is concerned, the amplifier took in stride everything we could throw at it in the way of heavy use and/or abuse. Its construction appears to be distinctly better than that of the Hafler DH-101 preamp. We foresee no problems.

What else is there to say? We're more impressed that we ever figured we could be by an inexpensive product and predict that the amplifier business will never be the same again.

Rappaport AMP-1
(follow-up)

A. S. Rappaport Co., Inc., Box 52, 530 Main Street, Armonk, NY 10504. Model AMP-1 stereo power amplifier, $1800. Three-year warranty. Tested early production samples, owned by The Audio Critic.

In our long introductory review of the AMP-1 in the last issue, we covered the basic points of the design in some detail but could evaluate only the earliest prototypes as to laboratory performance and listening quality. We're now able to report that the production version represents a significant step forward sonically, well beyond what we expected. The production Rappaport sounds cleaner, more transparent, more detailed, more neutral, more "unelectronic" than all other power amplifiers we're presently aware of, regardless of price or origin. No exceptions, no qualifications. Switching to any other power amp, even as good as our previous top choices, reveals tiny but discernible flaws that might have gone unnoticed without the direct comparison. As a consequence, the Rappaport AMP-1 is now our exclusive "Reference A" power amplifier.

That said, we must add that our reservations are legion. Although the production version doesn't reach, let alone exceed, 100° C (212° F) in temperature on any part of the chassis as did the prototype, the problem hasn't been solved to our full satisfaction. The amplifier is still much too hot, especially with high line voltage (ours occasionally rises to 122 or even 123 volts). There can be no question of anything but open-shelf housing; children and pets must still be kept away from it. In at least one sample, hum was distinctly audible through an efficient speaker system, even with the inputs shorted, and we had considerably more trouble with the AMP-1 than with any other amplifier trying to avoid ground loops in our complex multichassis system. But that's the least of it.

Our main reservation is about operating reliability. We had one massive failure after another. Output transistors blew on three or four different occasions; some of the lower-level transistors in the circuit boards also went sour. Even when the amplifier didn't shut down, there was grossly excessive DC offset in one or both channels that needed fixing, again at least three or four times. Grounds opened up inside the amplifier, causing monstrous amounts of hum. And so on and so forth. Andy Rappaport was, needless to say, extremely prompt and cooperative in giving us the required service and repairs; in fact, our original two production models were completely replaced, and three of the four channels in our two replacement amplifiers are no longer the original ones, either.

The question is, was all this typical or a statistical freak? Andy Rappaport claims that out of 60 other production AMP-1's in the field, only 3 have given the slightest trouble, and one of these was the victim of a horrendously fouled-up AC power line in a store, leaving only 2 internally caused defects out of 60 units, a defective rate of 3.3%. That's generally considered acceptable in a totally new and different product. (We asked him to put these figures officially into writing for our files, but he hasn't done so as of press time.) Is it possible that our high line voltage, combined with our constant plugging and unplugging of both inputs and outputs, plus our propensity for clipping amplifiers with piano master tapes, etc., created a totally abnormal operational environment for this amplifier? Or is the no-feedback design of the AMP-1 a destabilizing influence on bias conditions and other device parameters, so that the amplifier tends to live dangerously? We haven't got the answer.

Since Andy Rappaport himself defended the no-feedback philosophy at considerable length in our State of the Art seminar and was critiqued in depth by all the other participants, we'll add only one little tidbit here on the subject. At 20 volts output into 8 ohms (i.e., 50 watts, which is just a hair over half power for an AMP-1 channel), we measured THD of the order of 0.6% at any frequency below 1 kHz, rising to approximately 0.8% in the neighborhood of 10 kHz and 1.15% at 20 kHz. How about that? Remember, this is the world's best-sounding amplifier. What would it sound like if 40 dB of feedback could be applied to reduce these figures to the vanishing point? Like a DB Systems amplifier maybe? There's food for thought.

It should also be added that an outrigger device for bridging the two channels of the AMP-1 will be available shortly. The resulting mono amplifier will be able to deliver over 54 volts rms into any load down to 3 ohms or so. Into 8 ohms, that's a 370-watt amplifier; into 4 ohms, a 740-watter. Landlords, beware.

As for our overall assessment of the Rappaport AMP-1, we're of two minds about it. We'd like to tell every red-blooded audiophile to rush out and buy one. We put our money where our mouth is and bought two. But what if you live in the middle of the prairie or up in the mountains and the damn thing conks out on you? Or suppose you have a curious three-year old who can't keep his hands off large black objects. Just don't say we didn't present both sides of the question. We have our priorities and you have yours.

Recommendations

The following reflects our sifting of a large number of highly touted power amps since the beginning of this series of tests, but doesn't include two important Mark Levinson candidates. One is the current version of the ML-2, which has been reported to us by reliable parties as distinctly superior to the early ML-2's we had tested and reviewed; the other is the new ML-3, a 200/200-watt (into 8 ohms) class AB monster amplifier with an unusually large power supply. We shall see.

Best-sounding power amplifier tested so far, regardless of price, but with numerous caveats (see review above): Rappaport AMP-1.

Close to the best at a spectacularly lower price (and one of the three or four best regardless of price): Hafler DH-200.

Diamond Styli for True High-Fidelity Reproduction

By Dr. Sao Zaw Win
Win Laboratories, Inc.

Stylus tip geometry, diamond quality, cantilever materials and other determinants of stylus performance, mostly foreign territory even to advanced audiophiles, are discussed by a distinguished technologist who "has been there."

Editor's Note: **Our obvious respect for Sao Win's comprehensive knowledge of the subject, which prompted us to solicit this article, should in no way be construed as an endorsement of his products, which are received in our laboratory with the same skepticism before testing as all others. Nor should it be assumed that the main thrust of our phono playback investigations is necessarily identical to his; he makes no reference here, for example, to the largely quantum mechanical model of the stylus/groove interface as perceived by researchers such as James V. White, Mitchell A. Cotter and the Teldec group, the purely practical consequences of which appear to be confirmed by our experience. (See also the seminar transcript in this issue.) His basic facts, in any event, speak for themselves.**

Despite numerous improvements over the years in mechanical and electronic equipment for the playback of phonograph discs, the sound produced still depends firstly on the pickup head and the stylus itself. Therefore, the physical properties of the stylus and the precision with which it is manufactured are of great importance.

For many years, the steel needle satisfied most requirements but suffered from disadvantages from the point of view of quality sound reproduction and wear. Then the osmium point was developed, followed by the sapphire. The advent of the vinyl long-playing disc increased the importance of obtaining styli of the hardest material that could take a high polish and have a very low coefficient of friction. Diamond was the choice, but not until 1955 could diamond styli be manufactured commercially on a significant scale.

The theory of stylus design.

Most performance deficiencies for the reproduction of discs arise because of the differences in action and nature between the cutting and reproducing styli. It has been more than a decade since low-mass diamond styli first came into practical use for sound reproduction, and while the demand for both spherical and elliptical styli has greatly increased, further improvements have been made both in stylus shape design and production. It is necessary to establish the reasons and understand why these improvements in stylus shape were introduced and why the aforesaid two types of stylus profile have poor potential performance.

First, the *spherical stylus*. It is so named because its basic conical shape is rounded at the tip into a spherical form. The radius of this spherical region is critical because the stylus must fit the groove walls perfectly. (See Figure

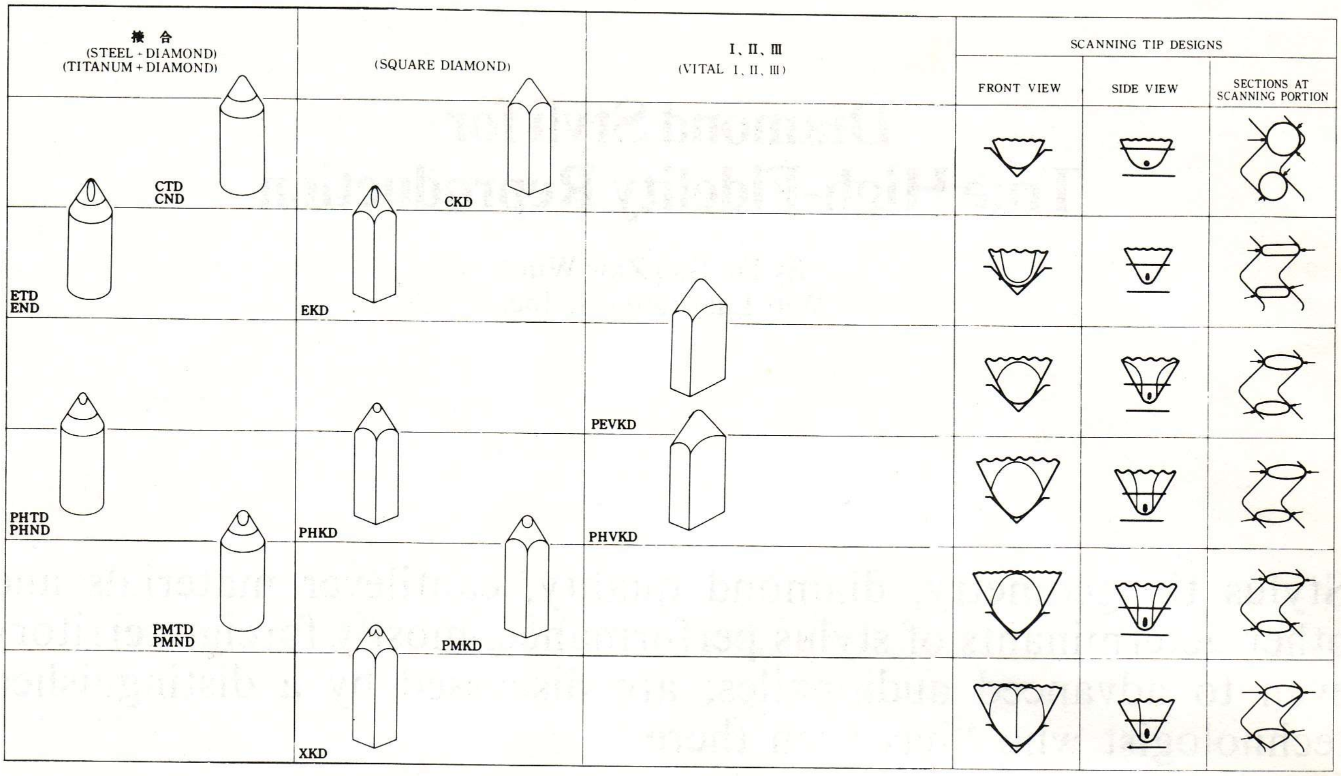


Figure 1: Various diamond stylus tip configurations.

Designation codes:

CTD, CND, CKD	conical point
ETD, END, EKD	elliptical point
PEVKD	Ogura point
PHTD, PHND, PHKD, PHVKD	Ogura point
PMTD, PMND, PMKD	Pathe Marconi
XKD	Shibata

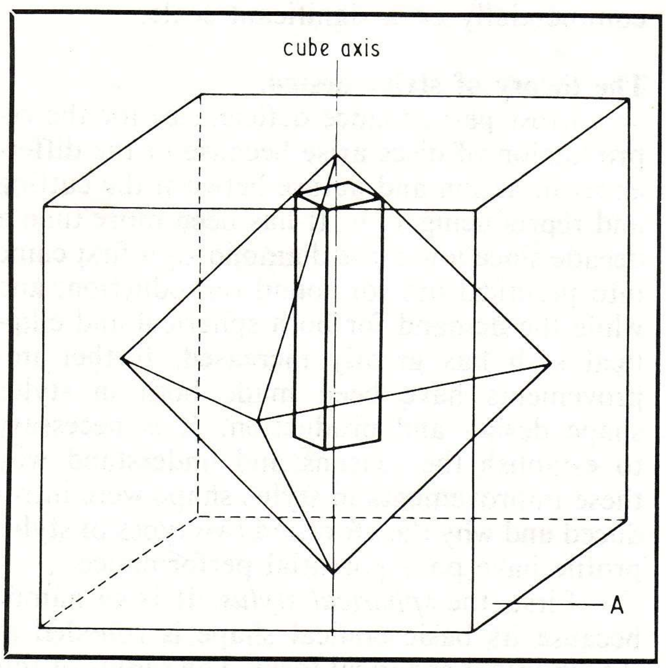


Figure 2: Orientation of cube, octahedron and rod.

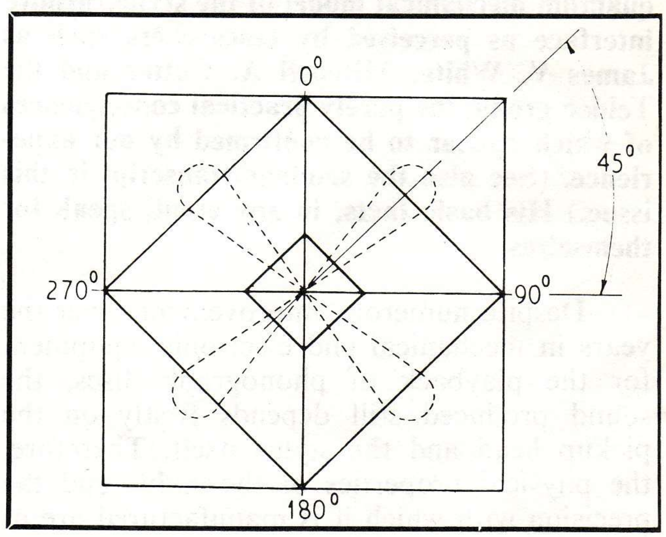


Figure 3: Orientation of cube, octahedron and rod, as seen normal to cube plane.

1, styli CTD, CND and CKD.) As can be seen, if the radius is too large, the stylus will ride along the top of the groove; if it is too small, the stylus will bottom in the groove. In either case, mistracking will occur; in addition, a bottoming stylus will become fouled with dust dug out from the normally untouched bottom of the groove, thereby increasing noise and distortion. Out of this critical fit two main problems arise: one is tracing distortion and the resultant pinch effect; the second is inner-groove distortion.

Tracing distortion occurs because of the way the distance between the groove walls varies with changes in the direction of the groove. This is the result of the flat chisel shape of the cutting stylus; when the deviation from nominal is considerable, the reproducing stylus is forced vertically upwards. Since a stereophonic pickup head is sensitive to vertical stylus motion, this movement due to what is normally called "pinch effect" causes distortion.

Inner-groove distortion is also directly linked to the nature of the spherically shaped tip. The outer grooves on the disc are longer than the ones closer to the center, so that the distance covered by the stylus in one revolution at the beginning of the record is far greater than toward the end. To produce modulations of a given frequency on any part of the disc, the cutting stylus must vibrate at a constant rate. However, near the center of the disc there is less space to accommodate the same rate of vibration and, as a result, high-frequency modulations in the inner grooves become so cramped that their radius of curvature may sometimes be smaller than the spherical tip radius. Therefore the stylus will no longer remain in contact with the groove walls and distortion is generated.

In order to eliminate the problem, a playback stylus as similar as possible to the cutting stylus was developed. This was the *elliptical stylus* (Figure 1, styli ETD, END and EKD). With its longer axis across the groove, it is far less susceptible to the problems of pinch effect, and the smaller ends are able to trace the tiny modulations at the inner grooves. There is a serious problem, however: because of the small contact area with the groove wall, the elliptical profile exerts greater pressure on the grooves and can cause damage. *(But see also the discussion of contact area and vertical tracking force in the seminar transcript.—Ed.)* Here "effective tip mass"—which is a combination of various factors such as the mass of the stylus, the mass of the cantilever, and the Young's modulus of the cantilever—plays an important part.

Technically we can assume that weight reduction of the diamond tip and of the cantilever reduces effective tip mass. In practice, however, reduction of stylus tip size is limited by the need to retain a particular tip configuration. There must also be a minimum size of diamond stock for operator handling in the manufacturing process, since orientation must be kept to close tolerances from the first step of production right down to the bonding of the stylus onto the cantilever.

With the advent of CD-4 discs and of the recent PCM and direct-to-disc recordings, the elliptical profile required a higher tracking force resulting in greater pressure on the groove walls. Unless the contact radius was at the same time limited to 7.6 microns, it would not trace the CD-4 carrier frequency fully. With an elliptical stylus of 7-micron side radius measured under 45° to the longitudinal axis, the high-frequency modulations of the groove wall are abraded because of the small contact area. The rate of record wear is also combined with a higher rate of stylus wear. For these reasons, new methods were developed to increase the contact radius and produce a tip profile which is very close to the shape of the cutting stylus, with an elongated zone in the contact area at a 90° included angle. (See Figure 1, styli having designations beginning with PE, PH and PM.)

This elongated zone is in practice limited by basic tolerances both in the groove itself (not all discs are cut with equal precision) and in cartridge/arm geometry (VTA, tone arm alignment, etc.); errors could cause the stylus to ride high near the top of the groove wall on one side, with the tip very low on the opposite wall. Research has shown that when all these parameters are given due consideration and everything is properly aligned, a long radius of 70 microns with a center 24 microns away from the main axis will give excellent results if the short radius is not smaller than 6 microns. Under 6 microns, with a nominal tracking load of about 2 grams, the abrasion starts again. The upper limit must be between 8 and 9 microns on account of the elasticity of most vinyl mixes, which is approximately 0.376 x

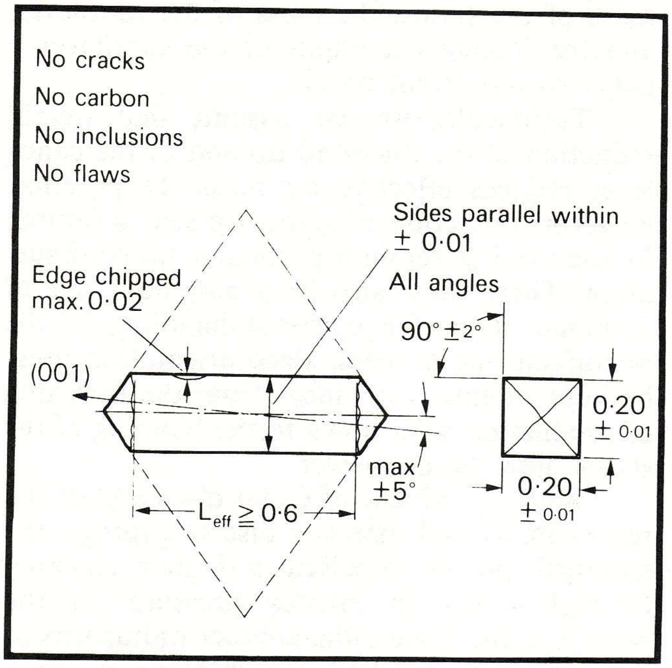


Figure 4: Specifications of oriented square rods for stylus fabrication.

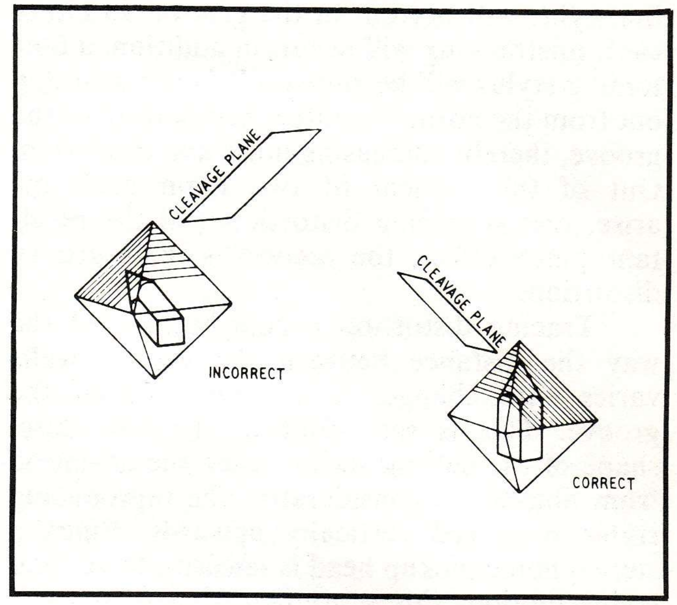


Figure 6: The two extremes of orientation of the rod within the octahedron. (Left) The worst orientation; (right) the best orientation.

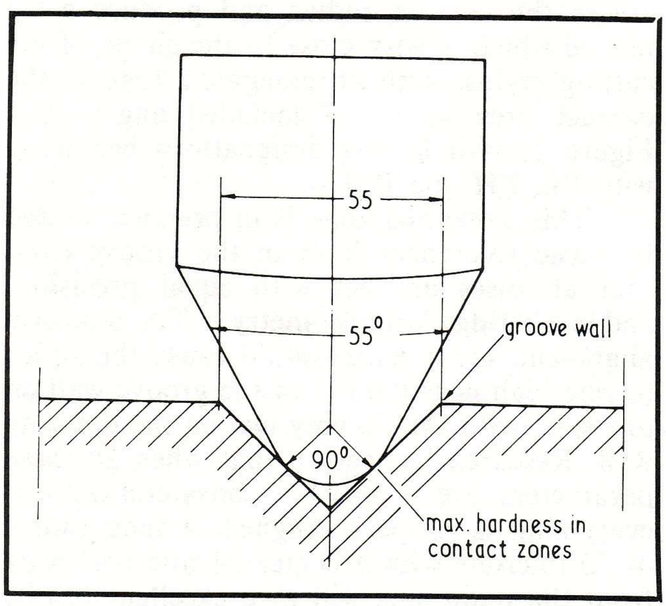


Figure 5: Stylus in correct orientation with respect to groove walls.

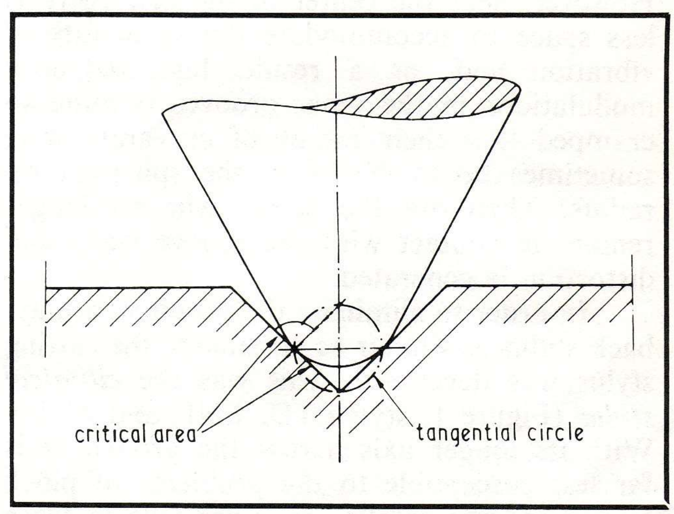


Figure 8: The tangential area of the stylus in relation to the groove wall.

10^{-10} cm/dyne.

Figure 1 also shows a *Shibata stylus* patented by the Victor Company of Japan (see stylus XKD). The shape is very close to that of the cutting stylus; the parabolic front side is created by two facets intersecting a polished cone, and the blending in of the sharp corners creates a small uniform radius over the whole contact area. But, the major radius being large, this stylus profile extends too dangerously close to the bottom of the groove; hence the possibility of mistracking as a result of the tip bottoming and getting pinched out of the groove under high modulation.

In principle, all these cuts—the *Shibata*, the Bang & Olufsen *Pramanik*, Ortofon's *Fine Line*, Shure's *Hyperelliptical* and ADC's *Aliptic*—stem from the tip shape that was patented for Pathe Marconi in 1954, the only differences being that, to create this shape in the contact area, different methods are applied by using flat facets, rounded facets, cones, etc. (See Figure 1, stylus PMKD.)

From all of the above explanations it now follows that an ideal version of the stylus must be cut or generated without facets. It could be described as an intersection of two cones with blended corners. While close to all of the above styli in configuration and design, this ideal version must also have smaller contact radii of 50 microns and 6 microns, ensuring not only that the contact area will be extended beyond what is normally attained by other designs but also that the small groove modulations will be tracked successfully. By also extending the area of contact upwards, so that the greater part of the stylus below the nominal disc surface is actually in full contact with the groove wall, this configuration applies less pressure and at the same time avoids the major problem common to all CD-4 styli, which have tips extending dangerously close to the bottom of the groove. Such an ideal stylus tip configuration actually exists and is called *Paroc* by Weinz of Germany and *Vital* by Ogura of Japan.

This type of cut has only one disadvantage: since the oval cross section cannot be detected under low magnification in a microscope such as is needed to handle the stylus, mounting on the cantilever becomes very expensive and difficult. Special provisions and tooling have to be made.

Diamond selection and fabrication.

Diamond styli are produced in three different qualities, which depend upon the size and nature of the natural diamonds. Three diamond sizes are used: (1) well-formed, flawless octahedrons; (2) small, 500-per-carat stones selected from grit; and for the cheapest styli (3) tiny, 1000-per-carat diamonds. (A carat is a unit of weight for precious stones equal to 200 milligrams.) The highest-quality styli are manufactured from octahedrons whose size depends upon the length of the stylus shank required. The initial stage is to prepare a rectangular rod of diamond from an octahedron. The orientation of the rod to the crystal symmetry is of extreme importance. The long axis of the rod must be within 3° of the cube axis, as seen in Figure 2, and the flat sides of the rod must be parallel to the octahedral edge (i.e., the 45° plane joining the two cube axes), as indicated in Figure 3.

Square-section blanks are produced in sizes ranging from 0.14 x 0.14 mm cross section by 0.50 mm length, up to 6 x 6 mm cross section by 8 mm length. The larger blanks are produced by sawing a rectangular rod from a well-formed, flawless octahedron. The starting point for the best styli consists of extra fine-quality selected natural crystals of size 100 to 400 per carat, with no chips, flaws, carbon or other inclusions. (See Figure 4.) Orientation of the longitudinal axis is normally parallel to one main axis of the diamond crystal, in other words from 4 point to 4 point or in the (001) plane in crystallographic terms. The orientation can be done by means of X-rays; however, on the smallest stones of about 0.2 mm diameter, the process takes several hours because of the low mass of the diamond to be penetrated by the X-rays and the low reflection density. This expensive orientation process ensures that the 4 point of the diamond will be well within the radius and that the 2 point (i.e., the 2-fold axis) will be normal to the groove wall when the record is played, thus presenting the hardest and most wear-resistant section to make contact. (See Figure 5.) The orientation of the 500-per-carat size is less easy, as these stones are so small that they cannot be processed into rods. Some orientation can be obtained, but it is not always possible to arrange that the point of the tip will be within 3° of the cube axis. The

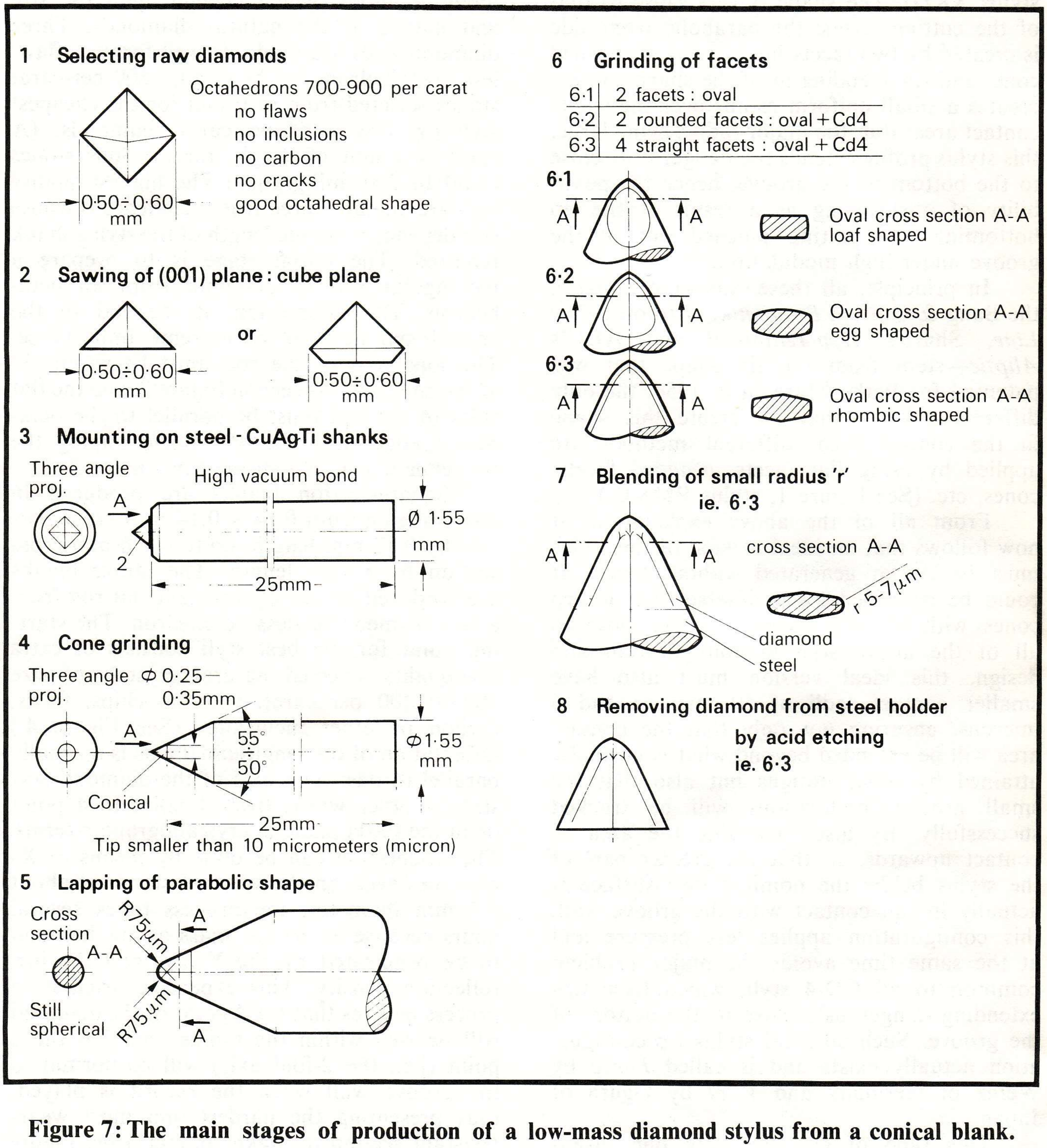


Figure 7: The main stages of production of a low-mass diamond stylus from a conical blank.

smallest crystals (1000 per carat), used in the cheapest styli, are simply tumbled to a usable size; since orientation is impossible, it is a matter of chance whether the hardest part of the tip will be in contact with the groove walls.

Orientation of the diamond is extremely important because when the hardest part of the diamond is in contact with the groove walls the wear on both the diamond and the disc is minimal. Deviation from this orientation reduces the life of both the diamond and the disc. The worst orientation is where the longitudinal axis of the diamond rod is normal to an octahedral plane. In such a case the octahedral planes will tend to cleave off and the diamond will act as a turning tool, destroying both itself and the groove walls. (See Figure 6.)

Figure 7 illustrates an alternate method, showing the typical stages in the production conical blanks for diamond styli of low mass. Stages 1 through 4 show that after the sawing of the cube plane the blank is bonded to a steel support (which is later removed) in order to facilitate polishing of the tip. The surface of the bonding area on the diamond is made rough and frosty to get a good bond to the steel support and later to the cantilever of the stylus assembly.

Bonding is done under high vacuum to eliminate oxidation and to achieve a good bond of diamond to metal. Stages 5 through 6 show the grinding and polishing of the stylus tip. The cone surface is fine-ground with a 600-mesh wheel and then polished with an 800 wheel. Cone grinding is done on special machines that take into account the crystallographic orientation of the diamond in order to minimize the effects of differential hardness, which could otherwise produce an oval cone. The polishing ensures the obtainment of a very sharp point, which is extremely important for subsequent grinding of the radius. The cone sides are not polished in the mass production of the cheaper styli, as only the radius at the very tip is involved in actual contact with the groove walls. (See Figure 8.) The diamond tip is then removed from the steel support by acid etching, so that the mass is considerably reduced. The weight of the conical blanks—and also of nonoriented ball-shaped blanks—after finishing as diamond styli is 0.2 x 10^{-5} gram, whereas the weights of cylindrical and square diamond rods after finishing are 1 x 10^{-5} and 1.4 x 10^{-5} gram respectively.

Cantilevers.

The cheapest version of a stylus cantilever is a conical aluminum tube. These are used mainly with bonded nonoriented tips. A more expensive cantilever is a low-mass full *beryllium rod* of 0.30 mm diameter. The advantage is that beryllium is of lower density than diamond (1.8 as against 3.5 grams per cubic centimeter).

Some ceramic substances such as alumina (Al_2O_3) and hot-pressed silicon nitride have been tried; however, their modulus of elasticity is not as good as of beryllium. The latest developments indicate that *boron*, with its favorable ratio of density or specific gravity to Young's modulus, is the best suited for cantilever material. Boron has the further advantage that it is less expensive than beryllium and is absolutely nontoxic (unlike beryllium, which is quite poisonous), so that operator handling is easier and potential health hazards are eliminated. (See table of comparison of material characteristics.)

Table of Comparison of Various Cantilever Material Characteristics

Material	Young's Modulus (x 10^8 dynes/cm^2)	Density (gm/cm^3)	Specific Rigidity (x 10^4 m^2/s^2)	Sound Velocity (m/s)
Boron	42,000	2.3	18,260	13,500
Beryllium	28,000	1.84	15,200	12,300
Carbon Fiber	16,200	1.42	11,400	10,700
Aluminum	7,400	2.69	2,750	5,200
Titanium	11,000	4.54	2,420	4,900

Bonding.

Normally, the diamond tip is bonded directly to the cantilever by means of high-strength epoxy resin. The big disadvantage is that the stylus starts to shift position as the epoxy hardens or polymerizes because of contraction.

Shrinking the aluminum or beryllium around the diamond is another possibility, but the absolute sizes are too small for obtaining a reasonable opening by cooling the diamond or heating the metal cantilever or both.

The latest technique is the bonding of diamond to boron, etc., under high vacuum by using a special silver solder. This must be done by wetting the diamond and the boron at the same time in the active bond with silver solder and titanium. Special tooling is used to prevent the displacement of the diamond stylus tip when the solder starts to flow, as surface tension pulls the diamond tip in all directions, thus tending to dislocate the diamond from the optimum position. This metallic bond is advantageous for its quick heat-sinking qualities, keeping the diamond stylus tip cool at high modulations. Also, no phase shift will occur under actual playing conditions as might happen with epoxy on acount of the latter's elasticity due to the acceleration of the stylus tip itself.

Conclusion.

The audio world has at least as good reasons to value the diamond, this amazing form of carbon, as jewelers do. Truly, the phono industry would grind to a halt in its absence. It is a good idea to take our cue from the best jewelers and refuse to consider anything but diamonds of the highest quality for our phono styli. A good phonograph record deserves the best diamond as well as optimum tip geometry and cantilever design.

Acknowledgements: **The author wishes to thank Dr. Ernst Weinz of Weka GmbH, Idar-Oberstein, West Germany, and Mr. Hiroshi Ogura of Ogura Jewel Company, Tokyo, Japan, for their technical inputs and valuable papers.**

FM Tuner Article Delayed

Our first tentative consideration of FM tuners, promised for this issue, will have to wait until the next one. We consider this to be a relatively low-priority subject, and something had to give to make room for all the other goodies in our fattest issue ever.

Cartridge/Arm/Turntable Developments

Part IV of our somewhat loosely connected series, this time focusing exclusively—more by coincidence than by design—on SOTA nominees and contenders. (Things won't look *always* this rosy.)

Before anything else, a quick apology. We regret that our promised turntable survey, with quantified comparisons of mechanically and acoustically induced sonic colorations in a variety of models, has been postponed until the next issue.

We don't feel terribly remiss about this; the measurements and evaluations involved are considerably more elaborate and time-consuming than in the case of, say, preamplifiers, and we simply aren't ready yet to publish a full report. What happened was that, as we got deeper and deeper into the subject, new complexities and ambiguities kept cropping up. As usual. We also discovered considerable overlaps with, as well as divergences from, similar studies done by Poul Ladegaard for Bruel & Kjaer in Denmark and by Martin Colloms for the Hi-Fi Choice Series in England, all of which we want to sort out thoroughly before committing our conclusions to paper. This is relatively unexplored territory and, as we've said before, it's hard to schedule in advance the solution of unsolved problems.

Don't expect, however, any radical contradictions of our previous opinions and recommendations when the final results are published. As we go to press, it still appears from our continuing tests that mechanical resonances and acoustical breakthrough are of far greater importance in explaining audible differences in performance than boiler-plate specs like wow, flutter and rumble, which are almost invariably subthreshold. Furthermore, sheer inertial mass and heroic measures of acoustical deadening, a la Cotter B-1, still appear to be unequaled by lighter-handed approaches when it comes to protecting the phono signal from all intrusions of extraneously generated energy. So don't look for any miracles from the latest lightweight, resonant tin can standing on four little rubber nipples.

Bad news from the VTA front.

We have only one piece of theoretical information to discuss here before the reviews, especially since the seminar transcript elsewhere in this issue isn't exactly light on theory. But this particular question is important and didn't come up in the seminar.

It seems that tuning the vertical tracking angle (VTA) by raising or lowering the back of the tone arm, or by shimming the cartridge, can under most circumstances yield only an advantageous trade-off rather than absolutely correct geometry across the board. (See also the letter to the Editor by F. Brock Fuller of the California Institute of Technology in our "Box 392" column up front.) The reason is that the only correct orientation of the long and narrow contact area on a modern stylus is straight up and down. That's because the cutting edge of the cutter stylus is also oriented straight up and down on the wall of the groove as the latter is being cut, even though the vertical *motion* of the cutter stylus is at an angle to the perpendicular. If you then compensate for VTA error by tilting the pickup stylus tip and in effect angling the contact patches, you no longer have an optimized "scanning aperture" and there will be a scanning loss in playback. A few degrees of VTA correction, while far from trivial when it comes to getting rid of audible FM distortion, will cause relatively small scanning loss. But the alignment can't be perfect, by definition, unless both the VTA *and* the contact patch orientation are 100% correct. And that will be the case only when playing records that were cut with the same vertical angle as the inherent VTA of the pickup. In other words—not very often. An industry-wide VTA standard is the only complete answer.

Now, going a bit further, if you have one of those typical present-day pickups with a VTA of, say, 29 degrees and then try to tilt it back by 14 degrees so it will play a 15-degree cut correctly, you're in deep trouble with almost any kind of elongated contact patch. There will be a significant scanning loss, as well as a definite torque on the tilted stylus tip as the groove tries to "straighten it out." The latter effect can cause all sorts of whipping motion and ultrasonic activity, highly undesirable both from the standpoint of clean sound and of record wear. Of course, a 14-degree correction may be mechanically impossible to begin with, unless the bottom of the cartridge case slants upward in the rear. (See also Volume 1, Numbers 5 and 6.)

It must also be added that a spherical stylus tip, with its round contact patches that have no orientation as such, is immune to these trade-offs. You can tilt a ball-point stylus backwards and forwards without changing the scanning aperture; you'll just have equally limited tracing ability and signal-to-noise ratio in all positions. A cartridge with a built-in VTA of 30 or 32 degrees might therefore be, paradoxically, better off having a stylus with round contact patches; at least you can apply massive VTA correction without side effects. Not that two wrongs add up to a right; the optimum pickup design for present records remains one with a VTA in the 15 to 20 degree range and a line-contact stylus tip. The advantages of such a stylus will be retained with a few degrees of tilt for VTA correction from record to record.

New literature on VTA.

The village atheist who professes not to believe in any of these superstitions about VTA is apparently still with us. One such professor, for example, recently stated in an audio club newsletter (not a learned journal, to be sure) that "VTA is for the most part nonsense" and that he is convinced that "those who hear great changes with VTA are using arms that are too flimsy and resonant." (Like our FR-66s?) Luckily, for those who need further convincing or just want the most comprehensive information on the subject, there's some considerably more authoritative *new* writing on VTA just becoming available.

The monumental two-part paper by James V. White and Arthur J. Gust we've known about for some time is finally out; the March 1979 issue of the *Journal of the Audio Engineering Society* contains "Measurement of FM Distortion in Phonographs" and the April 1979 issue "Three FM Methods for Measuring Tracking Angles of Phono Pickups." Almost concurrently, there's a more popular explanation of some of the same matters by Jim White in the June 1979 *High Fidelity* titled "Tracking-Angle Error: A New Slant." (Seventeen years old is apparently new to the hi-fi slicks.) These articles provide extensive references, bibliographies and/or credits, the latter including the never excludable Mitch Cotter. Dr. White's *High Fidelity* article begins with the sentence: "The odds are better than 100 to 1 that, astonishing though this may seem, your phonograph's sound suffers unnecessarily from as much as 5% distortion due to vertical tracking angle error." And that 5% refers to a flutter type of distortion, which even the dead can hear.

We trust that pip-squeaks who don't possess sufficient knowledge to be a file clerk to some of these distinguished researchers will now think twice before mouthing off about VTA from the third barstool on the left.

Fidelity Research FR-7

Fidelity Research of America, PO Box 5242, Ventura, CA 93003. FR-7 "pure" moving-coil phono cartridge, $660. Tested sample on loan from manufacturer.

This high-priced new flagship of the FR line is obviously an all-out bid for SOTA, and it falls short of the mark not because of sophisticated generator and stylus considerations but as a result of simple negligence of basic phono geometry. Once again, the VTA is ridiculously large; correction by lowering the back of the arm is prevented by the "professional" plug-in head configuration which leaves insufficient clearance for this maneuver. The integrated plug-in headshell also prevents any kind of twisting for lateral tracking error correction, which the pickup happens to need, alas, in the FR tone arms. So the battle is lost on account of trivia.

The really tough design problems, on the other hand, are beautifully handled in the FR-7. The coreless silver coil and unique double magnet with four poles constitute a virtually ideal generator, with superior orthogonality and torsional damping characteristics, as well as very low impedance and high output. The naked diamond stylus with long and narrow contact patches also appears to be excellent, so in essence all is well except that the whole thing sits there cockeyed in the groove. What a shame.

The sonic outcome of all this is that the midrange, distinctly clearer than that of the FR-1 Mk 3F or any other moving-coil pickup known to us except the Koetsu, is still discernible and enjoyable through the messed-up geometry, but the highs just don't sound right—not quite clean enough. Surgical transplant into some more readily adjustable headshell arrangement would be the only solution, but it might make the total acquisition cost comparable to that of the Koetsu, so why bother?

In our judgment, the incomparably less costly FR-1 Mk 3F is still this company's best shot at the title so far.

Koetsu

Koetsu, Inc., Japan. Imported and distributed in the U.S.A. by Sumiko Incorporated, PO Box 5046, Berkeley, CA 94705, and by Specs Corp., 1238 Green Street, San Francisco, CA 94109. Koetsu moving-coil phono cartridge, $995 or $1000 (depending on dealer). Tested sample on loan from Specs Corp.

Here we go again: this is our new reference cartridge. Sorry about that, but the Koetsu is simply the best we've ever heard. Its sound is clearer, more focused, more detailed, more solid, more "live" than that of any other cartridge known to us. We wish it weren't so, as we find the retail price nothing short of obscene, but that doesn't change the quality or relative ranking of the product. (The price structure, in case you're interested, is $350 from manufacturer to importer, landed; $500 from importer to dealer; $1000 from dealer to consumer. Draw your own conclusions.)

We're told that the Koetsu was designed by the retired chief engineer of Supex, a gentleman by the name of Sugano, who makes each sample by hand in collaboration with his son and sells only about one out of three, the other two being weeded out in a fanatically stringent process of quality control. (Don't hold us to this story; we didn't hear it from Mr. Sugano himself.) Only a few dozen pieces have reached these shores so far. The cartridge is easily distinguished by its rectangular capsule of wood, shaped out of a single block with beautiful grain and looking like the epitome of Japanese handicraft.

The stylus cantilever is no chopstick, though; it's a highly sophisticated structure of boronized aluminum (boron for high Young's modulus and propagation velocity, aluminum for low Q); in addition, its longitudinal axis intersects the diamond shank much closer to the actual playing tip than is usual, resulting in considerably superior dynamic behavior. The contact areas are of the tall and narrow variety, of course. The generator mechanism is equally advanced in design (samarium-cobalt magnet, special pole pieces, very high flux density—the works), and the power output in nanowatts is quite magnificent. So there are reasons for the good sound.

Interestingly enough, the Koetsu doesn't measure dead flat; there's a gently rising response above 10 kHz, up 6 dB at 20 kHz. We heard absolutely no zip or zing as a result, though, proving once again that amplitude response isn't the decisive spec.

And that's not all. The Mitchell A. Cotter Company has come out with a "dedicated" version of their moving-coil pickup transformer, wound especially for the impedance characteristics of the Koetsu cartridge and not adaptable to others by restrapping. The model designation is MK-2L; the price, if and when you can get one, is $650. Ridiculous? Well, we hate to tell you this, but the cartridge sounds ever so slightly sweeter and more solid through this transformer than through the standard Cotter MK-2 with P strapping. The impedance match is right on the button instead of just in the correct range, and the last few dB of S/N ratio and dynamic range are therefore extracted from the Koetsu. It's the combination we have in our current Reference A system.

Of course, if you don't make your living with audio and your dollars or yen don't come from the petroleum business, all this may seem like outrageous overkill. We wouldn't be doing our job, though, if we didn't report that nothing else we know of is quite as good. Overpriced, yes; swindle, no.

Pyramid RW-1 and RW-2

Pyramid Loudspeaker Corporation, 131-15 Fowler Avenue, Flushing, NY 11355. Model RW-1 turntable weight, $75; Model RW-2, $60. Tested RW-1 sample on loan from manufacturer.

The RW-1 is a 1 kg (2.2 lb) weight, milled out of a single block of stainless steel in an aesthetically pleasing shape, beautifully polished, with a spindle hole in the center of its bottom side. It fits snugly on the spindle of your turntable and virtually welds the record to the turntable mat, so that the grooves move strictly in accordance with the dictates of the platter, without any mechanical freedom of their own (well, almost without any). In other words, the disc is mechanically grounded to a much larger mass. The resulting absence of the tiny gives and takes and shifts and vibrations that are inevitable with a less positively located disc does indeed make an audible difference; the sound is distinctly cleaner and better focused, at least in a system of reference quality.

The RW-1 is too heavy for any but the really high-torque turntables, such as the Technics SP-10 Mk II and other top-of-the-line direct-drive models; for most users the lighter RW-2, which is only 500 grams (1.1 lb), will be the recommended weight. Of course, the greater the hold-down force, the better, especially for flattening certain stubborn warps.

Two reservations must be added. One is that no turntable weight of this type can correct a concave warp, since the center of the disc is already depressed and the weight can't hold down the outer edge. The other is that almost any kind of homemade weight could probably do the same overall job at much lower cost, but the Pyramid RW-1 happens to be an object of irresistible visual and tactile appeal.

Supex SDX-1000

(preview)

Sumiko Incorporated, PO Box 5046, Berkeley, CA 94705. Supex SDX-1000 moving-coil phono cartridge, $500. Auditioned one sample on private loan.

Just before press time, we had a very brief opportunity to listen to this latest top-of-the-line Supex in our Reference A system. The lateral and vertical alignments were carefully performed, so we're convinced that what we heard was valid.

We can report that this is an excellent moving-coil cartridge but not quite as good as the Koetsu, to which the SDX-1000 is intended to be the half-priced answer, we're told. The Koetsu is comfortably superior in smoothness, transparency, resolution and signal-to-noise ratio, the latter because of its vastly greater output capability. The Supex might even get beaten in a runoff against the FR-1 Mk 3F, a comparison we unfortunately didn't have time for. We believe the FR is possibly a wee bit smoother.

That's all we can tell you until next time; we suspect even this short preview will be more than what you'll see elsewhere about this very new and far from negligible product.

Technics SP-10 Mk II

Technics by Panasonic, Panasonic Company, Division of Matsushita Electric Corporation of America, 1 Panasonic Way, Seacaucus, NJ 07094. SP-10 Mk II three-speed direct-drive turntable with SH-10E power unit, $800 (chassis only, without base). Two-year warranty; carry-in service. Tested #DA7728B016, on loan from manufacturer.

In the last issue we promised a final vote of preference between this turntable and the Denon DP-6000, as mounted on the Cotter B-1 base in each case. (Remember, in the price-no-object category we can't wholeheartedly recommend the base of any Japanese direct-drive turntable known to us.) The nod goes to the SP-10 Mk II, not because of sonic differences but because it's a thoroughly professional piece of gear, capable of tremendous torque, built like an anvil, and virtually immune to abuse. For example, the technically most sophisticated FM station in our area, WNCN, has them running all day and all year without any problems. By comparison, the DP-6000 appears somewhat fragile and fussy, although its tachometer system (magnetically recorded reference track read by a magnetic head) is perhaps more refined and accurate, even if susceptible to misalignment. The issue has become more or less academic, since the DP-6000 is now off the market. Its successor, the DP-7000, is certainly not less fragile or fussy; the much costlier new DP-80, on the other hand, has a huge motor and looks like Denon's idea of a real blockbuster to pit against the SP-10 Mk II, but we haven't had our hands on one so far.

Another advantage of the SP-10 Mk II is its third speed; even if you never play any 78's it's great for speeding up test records and for really brisk cleaning with a record brush. We also like the brute-force electromechanical brake; it stops that powerful motor with astonishing abruptness, unlike those sissy electronic brake systems. Unfortunately there's no speed adjustment; the quartz-crystal phase-locked control is engaged at all times. Tough if you have absolute pitch.

Our initial experiments with measurement techniques for the quantitative comparison of mechanical resonances and acoustical breakthrough seem to indicate that the SP-10 Mk II in the Cotter B-1 system is something like two orders of magnitude better than typical audiophile turntables on their factory bases. It's a difference you can easily hear, although the Denon-Cotter combination is just as good sonically. More about all that in the next issue, as promised.

It's possible on occasion to confuse the servo loop of the SP-10 Mk II with a new push-button command and initiate cogging. This is most likely to happen when shifting to a lower speed without going through stop. You'll notice it immediately because when that big motor cogs, it makes the whole turntable shake. The thing to do is to stop the platter and start again at the desired speed. That's all; once the servo locks to the correct speed, cogging will definitely not occur spontaneously. It's about as much of a nuisance as starting your car now and then with two attempts instead of one; we can live with it. It's also possible that some sort of fine-adjustment of the power unit would cure this tendency permanently. We haven't bothered.

Until further notice, then, the Technics SP-10 Mk II in the Cotter B-1 base is our reference turntable.

Win Laboratories SDC-10

(preview)

Win Laboratories, Inc., PO Box 332, Goleta, CA 93017. SDC-10 DC Servo Reference Turntable, $2000. Tested #3, on loan from manufacturer.

This absolutely magnificent-looking turntable, a veritable sculpture in polished metal and clear plastic, arrived too late to be evaluated in depth against the Cotter B-1 system with the SP-10 Mk II, which is the only standard we could possibly judge a new $2000 unit by. (The price of the Win may even go up to $2350, we're told, if a slightly different motor with higher torque is decided on for later production.) So you'll just have to wait for our long-delayed turntable survey in the next issue if you want nitty-gritty comparisons between the two. Meanwhile, a few general observations:

The SDC-10 is a belt-drive turntable; you could say it's the ultimate embodiment of the Linn/Thorens approach, with rigidly clamped motor and synchronously jiggling platter and arm on soft springs. The DC servo motor is controlled by a circuit housed in a separate cylindrical switchbox; the two speeds are vernier-adjustable. Although the platter itself is quite heavy (9½ lb), it's our initial impression that the overall mass of the SDC-10 is still too small to present a really high mechanical impedance to brutal extraneous excitation. (We made the same comment about the Linn-Sondek a couple of issues back.) You can't manhandle the top plate of the Win with quite the same abandon and impunity as that of the Cotter. But then it's almost instinctive to treat such a beauty with a lighter hand.

This is not to suggest that the isolation and acoustical deadening of the SDC-10 are anything less than outstandingly fine. We originally suspected that the monolithic leaded plastic material of the translucent base would be no match in deadness for the laminated structure used in the Cotter B-1, but our first tentative measurements of mechanical resonances and acoustical breakthrough indicate no such inferiority. We find that quite remarkable. A few little anomalies we detected in the Win still need some sorting out; they may well turn out to be artifacts of our initial test setup.

Our listening evaluations so far have involved some discrepancies of the apples vs. oranges variety, attributable to the use of different arms and cartridges. We'll reserve judgment until a more uniform procedure is established. In any event, we figured that a preliminary report about a new superstar is always better than no report at all—which is what would have been normal with this kind of lead time.

Win Laboratories SDT-10 Type IIC

(follow-up)

Win Laboratories, Inc., PO Box 332, Goleta, CA 93017. SDT-10 Type IIC semiconductor disc transducer, complete with power source module, $350 ($550 after June 1, 1979). SPG-10 passive volume control module, with wide faceplate for both units, $150. Tested samples on loan from manufacturer.

If you think this is our final, definitive review of the Win strain-gauge cartridge, we may have to disappoint you. Not that we haven't had enough time to find out all about it or that we have the slightest hesitation about our conclusions. It's just that the SDT-10 represents a new technology in a state of ceaseless evolution, difficult to pin down at any given moment. We're unaware of any recent three or four-month period during which the SDT-10 remained totally unaltered physically and sonically. Not because Dr. Win can't make up his mind but because he is a scientist and a seeker of truth who can't stand being aware of a possible improvement and then *not* incorporating it immediately in his product for hard-nosed commercial reasons. (He ought to take a few lessons in the expedient spacing of upgrades from Joe Grado.) So we must first of all make sure *which* SDT-10 we're talking about.

Type IIC is the very latest version as we go to press, possibly so new that your dealer may still have a few of the older Type II in stock. The C suffix stands for Cycoenea, a proprietary name for a new alumina-based material of which the beam structure holding the semiconductor strain-gauge element is made. The older Type II had a pure alumina beam. A further difference is that Type IIC has a stylus cantilever made of boron on a graphite core, whereas the immediately preceding Type II has a slightly different boron cantilever with a tungsten carbide core. And that's not all. Dr. Win buys line-contact diamond styli both from Weinz in Germany (Paroc tip) and from Ogura in Japan (Vital tip), and he supplied us with his magnificently crafted plug-in stylus assemblies utilizing each type. And that's still not all. He also sent us various experimental styli utilizing other cantilever materials besides boron. So, even if we aren't exactly confused, we've had a surfeit of input to say the least.

Let's try to keep the whole thing as simple as possible. We can recommend the dead-stock, garden-variety Type IIC without serious reservations; it's one of the best cartridges around. But it happens to be the second best Win cartridge we've auditioned so far; the first best was the older Type II with a titanium stylus cantilever and Vital tip, a configuration not available commercially because Dr. Win feels that titanium develops metal fatigue much too easily. And again, that's not all. We had to raise the back of the arm way, way up to compensate for the rather small VTA of this cartridge and—get this—we had to apply 4 grams of tracking force to help damp out ultrasonic activity in the stylus and make the output sound as good as possible. Set up this way, and with the Win SPG-10 passive volume control replacing the entire preamp, this particular nonstandard combination of cartridge and stylus sounded absolutely stunning. The highs were the cleanest and fastest imaginable, and the rest of the range was almost as good. Only a very slight woolliness and smearing in the upper bass and lower midrange made us rank this sound below that of the Koetsu through the Cotter front-end modules. No other competition is visible. The stock Type IIC is very close in overall performance but not quite as smooth and transparent. It, too, benefits from increased stylus force beyond the officially recommended 2 grams and needs lots of VTA correction in the plus direction.

Now, what does all this add up to? We have spoken to Dr. Win about our findings and he is in basic agreement with us on most of the essential points. He believes that further damping of acoustical activity in the cantilever is indeed desirable and feasible, but he insists that it can be achieved with boron and with a vertical tracking force of 2 grams. He also agrees that his current VTA (14½° in Type IIC at 2 grams) may be on the low side for modern records and should be increased slightly. He is even working on a totally new technique to eliminate all needle drag (longitudinal translation) effects, which in our opinion may be the reason for the slight remaining imperfections. So, as you can see, the Win cartridge is a continuing process rather than a frozen product.

We feel that under the circumstances the consumer simply can't demand greater finality. A strain-gauge transducer with this kind of gauge factor, resulting in this kind of S/N ratio and dynamic range, would have been considered a wild dream only a few years ago. The surprising thing is that it exists at all, not that it isn't 100% finished. And even unfinished it's better than almost everything else around.

What's our immediate recommendation? If you don't care about the cost, go the Koetsu/Cotter route. If you don't think you need the switching and control functions of a preamp, you could save a lot of money by choosing the Win SDT-10 with its almost 1-volt reference-level output. No preamp, with the possible exception of the Cotter, is as transparent as it ought to be, so you'll be way ahead on that count alone. And the latest SDT-10 approaches the performance of the top moving coils, preamp or no. The SPG-10 will take care of the volume control function without any problems. Of course, the longer you wait, the better the SDT-10 will get; past experience points inevitably to that conclusion. But even if you get one tomorrow, we don't think you'll regret it. It's a technological tour de force. (We don't like that last-minute $200 increase in price, though.)

Recommendations

As always, we implore you to read every word in the reviews instead of simplistically depending on these summarized ratings. You won't be able to make an intelligent choice if you go by the "box score" alone. A number of recommendations have changed since the last issue; some of them remain the same.

Best phono cartridge tested so far, regardless of price: Koetsu.

Close to the best at an incomparably lower price, though still expensive: Fidelity Research FR-1 Mk 3F.

Special situation for the experimenter who can do without a preamp: Win Laboratories SDT-10 Type IIC.

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

Best separate tone arm per dollar: Series 20 Model PA-1000.

Best turntable tested so far, regardless of price: Cotter B-1 system with specially adapted Technics SP-10 Mk II.

Best turntable per dollar: See Reference B discussion in this issue (this is rapidly shifting ground).

Three New Step-Up Devices for Moving-Coil Cartridges

By the Staff of
The Audio Critic

Two more pre-preamplifiers and a transformer assault the gates of SOTA and don't quite make it, but the cheapest of them is an awfully, awfully good buy.

The only way to rate a moving-coil pre-preamplifier or transformer is to connect the best possible MC cartridge to it and listen; bench tests don't tell the whole story, as we've had occasion to explain in the past. We gave these units an extremely thorough wringing out in the laboratory, but it was the Fidelity Research FR-1 Mk 3F cartridge and, especially, the Koetsu that showed us what was really going on. Black-box electronic tests (and what could be a simpler black box than a pre-preamp?) are still in their infancy when it comes to audio; anyone who tries to tell you otherwise hasn't done much listening of the right sort.

The system into which each of these units was inserted for A-B testing was our Reference A (see the updates in this issue); levels were quite carefully matched. Since we know that our conclusions will be upsetting to a number of people, we want to emphasize in advance that our confidence in the validity of these listening tests is particularly high; duplication of our results, however, is possible only with the most meticulous alignment of lateral and vertical tracking geometry, and with a reference speaker that doesn't slow down transients or create significant time dispersion.

Audio Standards MX-10A

Audio Standards Corporation (Division of Duntech Industries, Inc.), PO Drawer 2529, 302 S. Melendres, Las Cruces, NM 88001. MX-10A moving-coil pre-preamplifier, $350. Five-year warranty. Tested #190, on loan from manufacturer.

This is a tough one. On the laboratory bench, with test signals of all descriptions going in and CRT displays and/or meter readings coming out, the MX-10A is closer to "a straight wire with gain" than anything we've ever encountered in the world of audio electronics. Its grounded-gate FET circuit, derived from RF technology (the discipline its makers come from), is mind-bogglingly wideband, fast and linear; no electronic test we could throw at it gave results even a small notch below perfection. In desperation we attempted to zap it with deliberately vicious and unreal-world complex waveforms out of a new synthesizer we have that can generate just about any signal you can draw with a pencil, and the output of the pre-preamp remained literally indistinguishable from the input. Incredible!

And yet—the MX-10A just isn't the best MC step-up device known to us. Even the $120 Marcof PPA-1 sounds better in some ways, and the Cotter MK-2 transformer in every way. Not that the Audio Standards isn't an extremely high-quality unit. It's beautifully made, and a few years ago its sound would have been considered a small miracle—it's that good. But today our expectations are higher and our criteria unforgiving.

The comparison with the $495 Cotter MK-2 transformer (standard P strapping) is fascinating. The approximately 80-kHz bandwidth of the transformer appears severely limited next to that of the MX-10A, which is a 2-Hz-to-VHF type of device; the transformer also has measurable ringing up there just before it drops, plus all kinds of other out-of-band characteristics that are numerically inferior by a wide margin to those of the pre-preamp. About the only electrical spec that doesn't leave the MK-2 mercilessly stomped is equivalent input noise resistance, where the laws of nature are in the transformer's favor. But when it comes to music, the Audio Standards has a distinctly thicker, denser quality, with considerably poorer delineation of subtleties. The Cotter sounds better focused, more finely etched and detailed, airier, more buoyant in dynamics. A spiccato violin passage, for example, sounds like the real thing through the Cotter and like very good hi-fi through the Audio Standards.

Now why should this be? Obviously, out-of-band performance is irrelevant, unless its defects are such that they affect in-band performance deleteriously, which doesn't appear to be the case with the Cotter transformer. It would also seem that the two-dimensional displays of the lab bench don't possess the holistic resolving ability of the human hearing apparatus, at least not when it comes to ultimate sonic subtleties. (See also the seminar transcript in this issue.) It's possible that there are highly elusive propagation and/or field effects in the pre-preamp which are avoided in a completely passive and symmetrical device like the transformer. Damned if we know. All we can tell you is what we measured and what we heard. We can't even point a finger at feedback as the possible culprit, since the MX-10A has no inverse feedback loop.

Isn't it nice, though, that we've come to this kind of impasse instead of having to worry about hum, hiss, cross talk, loads of TIM, etc., as we used to not so long ago?

Fidelity Research FRT-5

Fidelity Research of America, PO Box 5242, Ventura, CA 93003. Model FRT-5 toroidal step-up transformer, $455. Tested #026078, on loan from distributor.

Designed expressly for the new FR-7 moving-coil pickup (reviewed elsewhere in this issue) but specified as suitable for any MC cartridge with an impedance between 3 and 10 ohms, this expensive transformer is rather a disappointment. Compared to the Cotter MK-2, it has severely limited frequency response, especially on the bottom end but also on top, and quite high distortion.

On music, the FRT-5 sounds peculiarly thick and somewhat unpleasant. It gives a definite feeling of information displacement and/or loss. We don't want to belabor the point, since FR is one of the more "with-it" outfits in nearly all matters phonographic, but this just won't satisfy the audio purist.

That said, it matters very little that the FRT-5 provides switching facilities for three tone arms plus a straight-wire bypass around the transformer. On first things it isn't first.

Marcof PPA-1

Marcof Electronics, 7509 Big Bend Boulevard, Webster Groves, MO 63119. PPA-1 moving-coil pre-preamplifier, $119.95. Two-year warranty. Tested #1397100132, owned by The Audio Critic.

This is wonderful news: a $120 MC pre-preamp that's quite acceptable even to the ultrapurist with the highest standards. Yes, in an A-B comparison with the Cotter transformer, the top end of the Marcof comes off as a wee bit zingy and its bass a little pudgy. The pre-preamp is also somewhat noisier, by definition. But its overall sound is extremely clean, open and well balanced, leaving relatively little to be desired. For example, the Audio

Standards MX-10A is decidedly less transparent, even if slightly sweeter and smoother—for three times as much money. The Marcof has all the earmarks of a giant killer.

Our laboratory measurements revealed no faults worth discussing, although the Marcof isn't nearly as wideband, fast and straight-wire-like as the Audio Standards. Who cares? In its very adequate passband it stays out of trouble just as successfully, maybe more so.

There's a small penalty of inconvenience to be paid for all this value. The PPA-1 is battery-operated; you have to turn it on and off like a flashlight. The batteries won't last as long as the AC from your wall outlet, and there's a big on/off transient you should guard against by having your preamp volume control turned all the way down. We have no other negatives to report.

Until further notice, then, the Marcof PPA-1 is our Reference B step-up device for MC cartridges. Just in time, too; the Cotter transformer has become much too expensive for anything but Reference A.

Recommendations

As we still believe in the overwhelming superiority of properly designed moving-coil cartridges over ordinary magnetics, these MC step-up devices aren't just accessories but essential links in the chain of our recommended components.

Best way to play moving-coil cartridges, regardless of total cost: Cotter MK-2 transformer.

Best MC step-up device per dollar: Marcof PPA-1 pre-preamp.

A Brief Note on Absolute Phase

When a trumpeter at a recording session blows into his mouthpiece, the first transient wave front emerging from the bell of his instrument, the initial attack, pushes the microphone diaphragm ***in****. It's a positive-going signal and should be reproduced by a loudspeaker diaphragm moving toward the listener —a* ***push****. Similarly, a singer taking a sharp breath initially sucks the microphone diaphragm* ***out*** *and creates a negative-going transient signal that should be reproduced by a* ***pull*** *of the speaker diaphragm. If these signals are reversed in polarity, making the speaker push when it should pull and vice versa, the perceived sound won't be* ***exactly*** *the same. There will be a subtle loss of realism.*

The audibility of "absolute phase" in music (not to be confused with stereo channel phasing!) has been known for a long time to sophisticated audio practitioners; in fact in the early vacuum-tube days it was an ironclad rule in the recording studio that there must be no phase-inverting stages anywhere in the recording and playback chain. This traditional piece of studio wisdom is now being rediscovered with wide-eyed wonder by assorted new audio gurus and cultists, who hail it as the invention of the wheel.

With the widespread use of multimike, multichannel, op-amp-console, mixed-down recording, the absolute-phase criterion has become meaningless. Not even the cleverest recording engineer knows what happens to a positive-going pulse through that maze of signal paths; even if he did, he might end up mixing his signal with inverted versions of itself on the same track.

With exceedingly simple recording techniques, however, such as are used by Mark Levinson, Proprius, the "new" Max Wilcox and a few others, there remains the possibility that the positive or negative-going character of a signal will be preserved intact. In that case an extra touch of realism can be added to the reproduction by experimenting with the plus-minus polarity of each channel, either by quickly reversing the speaker leads on each side by hand or having some kind of two-position switch in each channel. (Needless to say, it won't work with speaker systems that have the woofer pulling when the tweeter is pushing—or have any other driver out of phase.)

Try it. You'll hear it. The better-sounding of the two possible connections will be the one with absolute phase.

More and Better Preamplifiers (Again and Again)

By the Staff of
The Audio Critic

An overwhelmingly superior SOTA preamp, an old favorite updated to near-SOTA and best per dollar, and the cheapest of them all improved beyond the previous best buy—this is a fast track indeed. Maybe the fastest in all audio.

Because of the excruciatingly detailed coverage of preamplifier design theory in our State of the Art seminar, both in Part I of the transcript in this issue and in the coming Part II, we wish to avoid driving the subject into the ground and shall therefore forgo our usual preamble to the reviews.

Just one small gloating chortle before we get down to business. It looks like that silly IEC playback equalization change has been laughed out of existence by those who knew better all along and said so—including this journal. We haven't seen it come up in new designs lately. *Requiescat in pace.*

* * *

The following preamps were all evaluated on the basis of bench tests discussed in previous issues and by insertion into our Reference A system for A-B listening comparisons.

Audionics BT-2 Series II

Audionics, Inc., Suite 160, 10950 SW 5th Avenue, Beaverton, OR 97005. BT-2 Series II Preamplifier, $459 (with front-panel handles). Three-year warranty. Tested #02962, on loan from manufacturer.

This "improved" version of the BT-2 can be distinguished from its predecessor by its gold-plated phono inputs and main outputs, plus a pair of -12 dB secondary outputs. The principal design change is a new regulated power supply; the wiring harness is also somewhat different.

We have no idea what else they did to this basically very decent preamp, which was our best-per-dollar selection only two issues ago, but we don't like the new sound. The only fault of the original BT-2 was the lack of ultimate detail, but its basic sonic quality was very smooth, balanced and listenable. The Series II, on the other hand, sounds hard, edgy and zingy. It grates on the ear, regardless of output level.

Our laboratory tests revealed no correlation with this finding; everything looked very acceptable, including preequalized square waves through the phono stage. The RIAA equalization isn't perfect below 200 Hz, mainly on account of the retention of the nonsensical IEC roll-off feature from the older BT-2. In the new "filter defeat" position of the switch, a little bit of the roll-off still remains (-1.25 dB at 20 Hz). Above 1 kHz, the curve is right on the nose.

All in all, not one of this excellent company's best efforts.

CM 301

Audio International Inc., 3 Cole Place, Danbury, CT 06810. CM 301 FET Preamplifier, $279. Three-year warranty. Tested #376, on loan from manufacturer.

Here's a case where nothing seems to be quite right. CM Labs claims that the 301 uses state-of-the-art circuitry and insinuates that the low price is due merely to the simple, straight-line, minimum-control design. We, however, find both the sound and the measurable characteristics unacceptable.

Intolerably hot, piercing highs and a homogenized, poorly detailed texture characterize the sound of the CM 301. The bottom end is diffuse, and there's a general lack of coherency throughout.

On the test bench there are similarly disturbing anomalies. The phono stage appears to handle transients asymmetrically when driven hard and the high-level stage is prone to peculiarly abrupt hair-trigger clipping once it runs out of headroom (of which it has plenty, though). Could be the feedback blues. The RIAA equalization error is one of the worst we've seen lately: +1.0 dB at 30 Hz, 0 dB from 200 Hz to 1 kHz, +0.5 dB from 5 kHz all the way up to 20 kHz. In other words, the bass and treble controls are up, even though the CM 301 doesn't have any.

Need we say more? Next!

Cotter PSC-2 and CU-2

Mitchell A. Cotter Co., Inc., 35 Beechwood Avenue, Mount Vernon, NY 10553. PSC-2 Phono Signal Conditioner, $475; CU-2 Control Unit, $1350 (projected price of forthcoming production version); both with PW-2 Master Power Supply, $250. Five-year warranty. Tested PSC-2 #C2-128410 with PW-2 #P2-128449, owned by The Audio Critic, and limited-production "engineering model" of the CU-2.

The five modules of Mitch Cotter's monumental 3-k$ "front end"—which he calls System 2 and includes his moving-coil pickup transformer, phono stage, control unit, noise filter/buffer, and power supply—can now be all hooked up together and auditioned, which is what we've been doing for some time (see also the reference system article in this issue). We can confidently report that nothing we've ever listened to during our 30 years in audio is in the same class sonically. Mind you, we've owned some truly superior preamps—by Audio Research, Hegeman, Saul Marantz, Mark Levinson, Rappaport and others—and each time it seemed that further improvement would have to come from elsewhere in the chain because the preamp was just about perfect. Well, it wasn't. Little flaws and glitches and obscurations attributed to the pickup, the record, the speaker, and so forth, came from the preamp after all. Because now they're gone. There's nothing but uncanny quiet and total transparency, coupled with unlimited dynamics.

That said, we must issue a few words of caution. The Mitchell A. Cotter Company is a recently formed, small and not very heavily capitalized operation, still in the teething stage as a business and, in addition, scarred by its rather traumatic separation from Verion Audio. Production has been slow, painful and somewhat unpredictable, although four out of the five System 2 modules are now in their absolutely final form and being made in small quantities every day. That much we know. The CU-2 in its production version has yet to be seen; several dozen of the electrically identical but cosmetically primitive "engineering model" have been circulating for some months, however. The point is—don't get mad if your now aroused desire for this equipment can't be instantly gratified, even if you walk into the store with a bundle of cash. Especially, don't get mad at **The Audio Critic**! Don't write us letters. We have absolutely no control over the operation of this company. All we can tell you is that, if and when you do manage to take home a piece of Cotter equipment, it will perform exactly as claimed. What's more, it will be built like a battleship, with nothing but mil-spec parts in it, ready to survive on the moon if necessary. For example, nobody but nobody in consumer audio uses the kind of heavy-duty, broadcast-type step attenuator that controls the output of the CU-2. It must cost the manufacturer something like $100.

Since we've already reviewed the Cotter

transformer and noise filter/buffer, which have no counterpart in competitive preamps, let's just consider the PSC-2/CU-2/PW-2 combination, constituting what is traditionally considered a complete preamplifier. At just a little over $2000, it's not even the highest priced example of the breed, although it isn't exactly cheap. What it will immediately do upon insertion into your system—and your system had better be a very good one, with the pickup perfectly aligned laterally and vertically—is to make you wonder where everybody has been. If this is what the record *really* sounds like, then what did—and do—all those other preamp designers have in mind? Is it possible that nobody but Mitch Cotter is conversant with the ultimate realities of phono playback? One should hope not, but then what's going on? The fact is that we've already begun to notice some frustration, jealousy and resentment out there in guru country about this very subject. Just remember that we're merely reporting what our ears tell us, without the slightest concern for repercussions in the fraternity house of high-end audio.

According to Mitch Cotter, the unique clarity and unstrained headroom of his preamp are due to relentlessly straightforward applications of basic principles rather than any kind of engineering wizardry. The signal is amplified in the current mode, as it is less vulnerable to contamination that way than in the voltage mode, and there is no inverse feedback loop anywhere in the circuit. Balanced-line symmetry is maintained from input to output. The stage of gain in the CU-2 is identical to that in the PSC-2, the only difference being the passive RIAA equalization in the latter. The conversion of current back to voltage in each circuit module is totally unorthodox but it works very simply and beautifully. The CU-2 has an unusually complete range of controls, one of the most useful of which is 180° phase reversal, separately controllable in each channel. (See the note on absolute phase elsewhere in this issue.)

On the lab bench, the behavior of the entire system is flawless, requiring no specific comment. We can't trip it up with any test known to us. The RIAA equalization is 100% accurate.

There's really nothing else left to say. If you can afford the Cotter preamp, get it. If you can't, at least go out and listen to it in a good system. It's an education.

Dyna mod: FET-5 Mark V

Jensens Stereo Shop (Frank Van Alstine), 2202 River Hills Drive, Burnsville, MN 55337. FET-5 Mark V preamplifier, $399 (when built new from Dyna PAT-5 Bi-FET kit). Mod kits, updates of older PAT-5 mods, etc., also available. Tested mod of original PAT-5, owned by The Audio Critic.

Frank Van Alstine has been modifying Dyna PAT-5's and PAT-5 Bi-FET's since the earliest memory of man. This is the latest mod we've been able to test; undoubtedly, by the time we're in print, there will be a Mark VI. You can make book on it.

This particular mod bypasses the tone controls and a few other functions of the PAT-5 in a professed attempt to achieve state-of-the-art performance on straight-through phono playback. The attempt is unsuccessful. The FET-5 Mark V betrays obvious colorations from the very start when A-B-ed against top-quality preamps. In the midrange, especially, there's a strangely hooded quality. We must admit, however, that there's no edginess.

The midrange also looks peculiar on pre-equalized square waves through the phono stage. A 500 Hz square wave, for example, which should have an almost perfectly flat top in this test, shows a shallow S-shaped top instead. The RIAA equalization error curve is humped up +0.4 dB at 200 Hz and +0.3 dB at 10 kHz, which isn't state-of-the-art, either. There also seems to be some oscillation at a very high frequency on square waves through the high-level stage. Maybe the thing is too wideband (rise time 250 nanoseconds—what for?).

In this instance, we can't recommend Frank Van for Ludwig van.

Hafler DH-101
(Improved, with DH-102)

The David Hafler Company, 5817 Roosevelt Avenue, Pennsauken, NJ 08109. Model DH-101 Stereo Preamplifier, $299.95 wired. (In kit form, $199.95.) Model DH-102 Moving-Coil Pre-preamplifier, $74.95 (fully assembled, to be connected inside DH-101). One-year warranty. Tested factory-modified sample, on loan from manufacturer.

As of May 1, 1979, beginning with serial number 1919000, all Hafler DH-101 preamps shipped, whether kit or wired, incorporate certain electronic improvements. The sound is considerably more three-dimensional, transparent and detailed than before, with the occasional hardness and edginess of the original version significantly reduced. (The RIAA equalization peculiarities are the same as before, though.)

We marginally preferred this improved DH-101 to the Hegeman preamp reviewed in the last issue (not the greatly improved Hegeman reviewed below), which is high praise indeed for a $300 preamp. At one point we were strongly considering the improved Hafler as our new Reference B selection but ended up filling that slot at a much higher price level. Even so, this unit has class, and we recommend it highly.

We can't say the same for the neat little DH-102 pre-preamp that fits right inside the DH-101 and is nourished by the same power supply. It sounds much too bright, zippy and fatiguing to be our recommendation even for a minimal moving-coil phono system. Try the Marcof PPA-1 instead.

Hegeman HPR/CU
(Improved)

Hegeman Audio Products, Inc. (Hapi), 176 Linden Avenue, Glen Ridge, NJ 07028. Model HPR preamplifier with Model HCU control unit (incorporating power supply for HPR), also known as Hapi One, $720 complete. Two-year warranty. Tested #260/238, on loan from manufacturer.

This is a very short and sweet story. Stew Hegeman decided he wasn't 100% satisfied with the Hapi One preamp we've been talking about for the past two issues, took some gain out of the phono stage, added that much gain back to the high-level stage, made a few other minor circuit improvements, and changed the faceplates from black to silver. The result is a totally—and we mean totally—different preamp.

The improved Hapi One, at $720, is in our opinion the best-sounding phono preamp in the world after the $2000-plus Cotter and therefore the greatest preamp bargain in existence despite its still rather high price. In openness, solidity, precise spatial imaging and fuzzless definition it outdoes some of the most revered names in preamps, costing several times its price. Its edgeless highs and tight bass are most impressive, and only when A-B-ed against the incredibly transparent Cotter does it appear the least bit veiled. The $1300 Rappaport PRE-3 also has a more transparent high-level, but not phono, stage.

On the test bench, the basic characteristics of the preamp are as before. Note that the new silver faceplates also identify retrofitted samples, just as they do the new production model, because whenever the innards are updated on an older unit the faceplates are automatically changed.

And that's not all. The Hapi Two, with the identical circuitry and controls but stunning new all-in-one cosmetics, is just about to make its debut. Its ultraflat "pancake" relay-rack chassis and stylish front panel are as sleekly glamorous as the Hapi One is drab and utilitarian. Clever shoehorning has made the previous two-chassis construction unnecessary, we're told; the hum is even lower than before. The projected price is $900.

So, 30 years after the marvelous but shockingly ugly Lowther-Hegeman horn speaker, it looks like Stew Hegeman has his whole act together.

Hitachi HCA-7500

Hitachi Sales Corporation of America, 401 West Artesia Boulevard, Compton, CA 90220. Model HCA-7500 stereo preamplifier, $370. Tested #8001053 G, on loan from dealer.

A young American manufacturer of esoteric audio components looked at this preamp on our test rack and shook his head: "I don't know how they do it."

The HCA-7500 looks like $800's worth of equipment at the very least, with its highly authoritative black control panel, solid-feeling knobs and switches, and mind-blowing variety of functions. And for a $370 preamp it sounds great, too. There's just a touch of zippiness in the midrange and highs, quite a bit short of an objectionable edge, and the overall sound is strikingly open and detailed. For a while we hesitated between the HCA-7500 and the newly improved Hafler DH-101; we finally decided that the latter with its somewhat smoother highs would be the better choice for the audio purist on a budget, even though it sounds slightly "rounded off" in detail next to the Hitachi.

Our laboratory tests revealed no vices; we were particularly impressed with the virtually perfect RIAA equalization. If this is a foretaste of what is to come from the Japanese giants as they become more interested in the sophisticated audiophile market, some of the small American firms that cater to the latter had better look to their laurels. An outfit like Hitachi is capable of doing just about anything they choose to do and package it at just about any price they feel like.

Mark Levinson LNP-2

Mark Levinson Audio Systems, Ltd., PO Box 6183, Hamden, CT 06517. LNP-2 Preamplifier, $3500 (plus special options, if any). Five-year warranty; customer pays all freight. Tested #2018, owned by The Audio Critic.

We own this superbly built, virtually indestructible professional control unit as a necessary extension of our Mark Levinson ML-5 master recorder system, which uses the Studer A80 tape deck and MLAS electronics. We wouldn't dream of owning it, for purely audiophiliac purposes, as the phono preamp of some kind of super system. Since a number of audiophiles to whom price is no object might be tempted to do exactly that, we feel the obligation to evaluate the LNP-2 very briefly on the basis of its essential sonic quality, without reviewing it at this time as a tool for the recordist. (This issue is already much too fat for that.)

Basically, the LNP-2 has a very clean, firm, open and well-controlled sound, without any irritating edge. A couple of years ago this would have been considered definitely SOTA. Next to the Cotter, Hegeman, Precision Fidelity C4 and Rappaport PRE-3, however, the LNP-2 appears to have slight colorations as well as an overall "homogenized" quality that submerges ultimate detail. This is true of both the phono stage and the high-level section; in fact we were quite surprised that the latest Hegeman HCU control unit, which isn't quite straight-wire-like either, changed the sound of a line-level source a lot less on a bypass test than the back end of the LNP-2. Time marches on.

As a result of these findings, we decided not to make the LNP-2 the nerve center of our reference system through which line-level signals would be routed. Too damn bad, because with its meters and calibrated controls it would have been ideal for that job.

Precision Fidelity C4
(Improved)

Precision Fidelity, 1238 Green Street, San Francisco, CA 94109. C4 dual-cascode preamplifier, $1095. Retrofitting original unit to new configuration, $75. Three-year warranty (tubes one year). Tested retrofitted sample, on loan from manufacturer.

Beginning with serial number 4000, the C4 vacuum-tube preamp comes with 10 dB less feedback in the phono stage, 10 dB less feedback in the high-level stage, an improved input selector switch, and other minor changes. The sound is even more open, detailed and dynamically alive than before, but meanwhile other good things have happened, and we can no longer make the C4 our Reference A choice.

The new Cotter preamp is in a totally different class, and even the considerably less costly Hegeman, in its latest version, sounds smoother on top and firmer in the bass. By comparison, the highs of the C4 have a tiny bit of shimmery coloration and the lows are on the loosey side. But, we must repeat, only by comparison against the best. The C4 is still one hell of a preamp.

We find it regrettable, though, that Precision Fidelity didn't use the opportunity of the design update to correct the saddle-and-hump error curve of the RIAA equalization. In fact, it's a little saddlier and humpier than before: -0.6 dB at 175 Hz (in the worse channel) and +0.35 dB at 10 kHz. Despite that, if you bought a C4 on the basis of our recommendation, we feel you should spend the $75 the manufacturer charges for the retrofitting. It's definitely worthwhile.

Rappaport PRE-3

A.S. Rappaport Co., Inc., Box 52, 530 Main Street, Armonk, NY 10504. Model PRE-3 Stereo Preamplifier (with external power supply), $1300. Three-year warranty; manufacturer pays all freight. Tested early production sample, on loan from manufacturer.

After our rather enthusiastic preview of the PRE-3 in the last issue, it will probably strike you as anticlimactic that, even though we find the sound of the production version quite excellent, we just can't live with that noisy phono stage.

Since we didn't have the early prototype side by side with the production model, we can't tell you whether the latter is a little quieter, as had been promised. All we can tell you is that it isn't quiet enough. The hiss intrudes at all times and submerges delicate details of texture and dimensionality. We found ourselves unable to listen "through" it undisturbed; to the extent we could do that, we found the sound of the phono stage very good but not as transparent or revealing, we felt, as that of the Cotter PSC-2 or even as satisfying overall as that of the Hegeman HPR. But we certainly can't defend too vigorously any perception based on such an artificial separation of subjective criteria.

The technical reason for the hiss is passive bandwidth limiting right at the phono input, before the first active stage; all the high-frequency noise of the cascaded active stages comes out the back of the preamp unimpeded, right on top of the phono signal. Andy Rappaport believes that this is the only way to implement his basic concept of a no-feedback preamp and that the issue isn't negotiable. To us that's something of an orthodox Marxist attitude; the political dogma isn't negotiable even when the net result is bread lines. Or, in this case, noise.

To complicate the matter, the RIAA equalization of the PRE-3 is quite inaccurate; there's a -0.5 dB saddle at 150 Hz, a +0.5 dB hump at 4 kHz, a -1.0 dip at 20 kHz, continuing to -2.5 dB at 43 kHz. That nudges the threshold of audible colorations. Harmonic distortion is a whole order of magnitude higher than in conventional feedback phono stages, but that doesn't bother us in the least. (See also our review of the Rappaport power amp in this issue.)

We must add that none of the above criticism applies to the perfectly quiet high-level stage of the PRE-3, which approaches straight-wire transparency on a bypass test and must be considered in the same class with the Cotter CU-2. But not many people buy a phono preamp for its "aux" sound.

We're hoping that one day Andy's dogmatism will mellow and he'll redesign that phono stage. Then the PRE-3 will be a killer.

Spatial Model TVA-1

Spatial, Inc., 3633 Long Beach Boulevard, Suite C, Long Beach, CA 90807. Spatial Coherence Preamplifier, Model TVA-1, $1195. Three-year warranty; customer pays all freight. Tested #00291 and #00496, on loan from manufacturer.

We liked the original version of the Spatial well enough to have used it as our temporary Reference A preamp while our Precision Fidelity C4 was being retrofitted and before the arrival of the latest Cotter and Hegeman units. We thought the highs were very suave and finely detailed, and the overall sound quite realistic in openness, depth and three-dimensional imaging. At the same time we noticed a somewhat thick, gagged quality in the midrange, which bothered us from time to time. We then switched to the Mark Levinson LNP-2, the only other good preamp we had available at the time, and the midrange opened up but the texture became somewhat more homogenized.

The company then informed us of a major sonic improvement and swapped our unit for one of the new ones. The midrange of the latter sounded considerably thicker and more stuffed up; if the first one had a wiener in its mouth, then the new one had a hero sandwich or a whole salami. This wasn't even close to reference quality. Spatial's explanation was that a few units had certain elusive problems originating from the new high-intensity neon pilot light; if we disconnected it the midrange would clear up. This sounded like sheer cultist dementia to us but we tried it anyway. Since the chassis is deliberately designed like a Chinese puzzle to prevent people from taking a peek inside (more about that in a moment), we had to take off and later replace all the eight knobs on the unit plus countless screws in order to get at the bulb. You guessed it—the sound remained unchanged, and we felt like someone who had knowingly sat down on the whoopee cushion planted by the kids, just to let them have their little joke.

On the lab bench both samples behaved in exemplary fashion, with very low distortion figures even at unusually high output levels, excellent square wave response both through phono and high-level, and accurate RIAA equalization. No clue whatsoever to the sonic anomalies.

Of course, anything but the most obviously audible superiority to all other preamps in the world would be incompatible with the introductory ballyhoo about the Knapp TFET-Valve used in the Spatial. This is supposed to be a revolutionary device that's more linear than any vacuum tube, bipolar transistor, field effect transistor or anything else that amplifies. But what is it? Ah, they won't tell you. They don't even want you to look; that's why the preamp is fitted and screwed together the way it is and sealed with an initialed tab. If you do succeed in laying eyes on the forbidden sight, you'll instantly think of the Audio Research "Analog Module," a hype of almost equal effrontery. The TFET-Valve is the same kind of potted circuit module—just a small block of epoxy. The words "patent pending" appear only once as a footnote in all the Spatial literature that mentions the TFET-Valve, and even there in a different typeface, tacked on as an obvious afterthought. No one we know has ever seen or even heard of any written material on such a patent. No one knows what's buried in the epoxy. No other electronic product in the world uses the device to our knowledge. Mighty peculiar for a revolutionary invention.

A half-technical, half-popular treatise by Richard P. Knapp on "spatial coherence and the TFET-Valve amplification process" was circulated long before the introduction of the preamp; it's an obscurantist, evasive piece of writing that couches familiar knowledge about time-dispersive distortions in the less familiar language of information theory and other disciplines. It's basically hamburger served as "Salisbury steak" by someone who doesn't want you to know it's hamburger.

All this devious manipulativeness and cultist posturing, combined with an apparent dismissal of the consumer's ability to think, cannot fail in the end to turn off the best people in audio, the very element that constitutes Spatial's potential following. We're already beginning to see some evidence of that. Actually, with all the preamp's faults, we like it a lot better as a product than we do the company's image itself.

Recommendations

Two issues ago, we made some remarks here about not expecting any changes in our top recommendations in the foreseeable future. And you know what happened in the very next issue. This time we'll be smarter and just name our current choices.

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

Best preamplifier per dollar (and, incidentally, the next best regardless of price): Hegeman HPR/CU (Hapi One).

Our Big and Our Little Reference Systems, Updated

Reference A, our experimental and not exactly practical five-figure super system, has changed only slightly. Our best-sound-per-dollar Reference B, however, is totally different, thanks to exciting new developments in medium-priced components.

For the detailed rationales behind these two different reference systems, please go back to the original article in the last issue. Here we merely wish to reiterate a very basic point of view we've been expounding since the earliest days of **The Audio Critic**, namely that only two choices of equipment are of genuine interest to the serious audiophile: (A) the best in sonic performance, regardless of price or other considerations, and (B) something reasonably close to the best, at a much, much lower price—if such a thing exists. Thus, the world's third-best preamplifier over $1000, or the fourth-best under $500, is an absolute bore even if it happens to be a respectable engineering achievement and the designer's mother is proud of it. The fact is that only Reference A and Reference B, conceptually speaking, are worth considering at any given time in any given component category, unless some very specific reason exists for a substitution.

That doesn't mean, of course, that if you own our last issue's Reference B power amplifier, for example, you should now throw it away as a piece of junk. It's still every bit as good as it was when we recommended it. But time marches on, and the Empire State Building isn't the tallest in the world anymore. If you feel that your present equipment is no longer enjoyable because it hasn't been blessed in **The Audio Critic**'s latest updates, you've got a problem.

We must add that the extensive changes in Reference B are quite unusual and unexpected. We really don't believe that the same thing will recur in the next issue. It's just that progress occasionally comes in quantum jumps. The gradual evolution of Reference A is much more typical.

Reference A

We must repeat once more that this is not for the well-heeled amateur without test equipment and a complete understanding of what's going on inside each component. Don't blame us if you rush out to buy all this stuff and then end up with problems. It's infinitely safer, and in most cases considerably more rewarding, to put all the responsibility squarely in Mark Levinson's or Harold Beveridge's lap, especially for the "back end" of your super system. Let them worry about your needs and holler at them if you aren't satisfied. Reference A is simply our way of saying: this is what's possible today with ready-made components and this is what we're currently using in our listening evaluations. It's a tool rather than a take-home package for the consumer. Okay?

Speaker System

Two out of three components remain unchanged here, and even the third one is still the same brand, though not the same model.

The *tweeter* is the Pyramid Model T-1 ribbon as before, but in the improved version (see follow-up review in this issue). The price has gone up to $1175 the pair.

The *midrange* speaker is now the Koss Model One/A electrostatic, which has considerably more headroom than the Model Two and, unlike the latter, doesn't have any panels driven out of phase. The price is $3000 the pair, alas.

The *woofer* remains the Janis W-1, at $1350 the pair.

The tweeter and midrange must be geometrically aligned for pulse coherence and the rearward radiation of the Koss must be blocked with sound-absorbent material, such as Tuflex.

Power Amps and Crossovers

The Pyramid T-1 is driven by a separate Rappaport AMP-1 ($1800) this time, with the tweeter's built-in high-pass filter/attenuator acting as a 2.5 kHz crossover.

The Koss One/A is driven by another Rappaport AMP-1 ($1800); we're currently working on an internal modification of the Koss to make it roll off by itself approximately where the Pyramid cuts in, but meanwhile we're feeding the power amp from the low-pass output of a Symmetry ACS-1 active crossover ($650), set for 2.5 kHz. The high-pass output of the ACS-1 isn't used.

The Janis woofer is still driven by the Janis Interphase 1 bass amplifier ($495 each, two needed for a pair of woofers). The built-in 100 Hz electronic crossover of the Interphase takes care of the bass/midrange split.

Preamplifier and Interfaces

We're quite decisively sold at this point on the Cotter front-end modules, despite the shocking price increases. (See Cotter preamp review in this issue.) The PW-2 power supply ($250) provides four identical power sockets for our four separate modules: the PSC-2 phono stage ($475), the CU-2 high-level stage with controls (so far available only in custom-built "engineering models" but soon to be in production at around $1350), and two NFB-2 noise filter/buffers ($425 each).

We use one NFB-2 at the input of the Janis Interphase 1 and the other at the input of the tweeter amplifier, which is driven full range; the midrange amplifier doesn't need one since its passband in this system is only 100 Hz to 2.5 kHz. There can be no doubt that the non-fatiguing, "zingless" sound of Reference A is at least partly due to this time-domain corrected filtering; out-of-band garbage is dumped overboard without altering the in-band information. Removing the Cotter filter/buffers makes the system sound just a little more like hi-fi and a little less like music.

Phono Cartridge and Transformer

This is a different combination than before; we're now using the Koetsu moving-coil cartridge (approx. $1000) with the Cotter transformer especially made for it, the MK-2L ($650). The resulting sonic improvement is instantly audible even if it isn't dramatic. (See review in this issue.)

Tone Arm

We still haven't found anything we prefer to the Fidelity Research FR-66s twelve-inch arm ($1250), although we do wish its VTA adjustment facility during play covered a spread of more than just 2° or so. Let's be thankful for small favors, though; a 2° range allows us to play most of the recent LP's correctly, without fiddling with the fixed arm pillar adjustment.

Turntable

We've settled down to the Technics SP-10 Mk II in the Cotter B-1 base (approx. $2100 assembled) as our reference standard until further notice. (See the turntable reviews in this issue.)

* * *

The total retail price of this revised Reference A comes to between $18,000 and $19,000, depending on the inclusion of the Symmetry crossover, small price breaks you can get on a few items, small extra charges here and there, etc. And that doesn't include an FM tuner or any kind of tape deck. But the sound is definitely better than before. As a matter of fact, it's very, very good.

Reference B

We really prefer to talk about this system because it's something you can go out and buy without a flirtation with insanity, whether fiscal or technophiliac, and enjoy immediately out of the cartons, like takeout fried chicken. The sound isn't in the same class with that of Reference A, but it's better than a lot of people have ever heard, especially now that we have new and improved component selections in nearly every category. These are all reviewed elsewhere in this issue.

Speaker System

Despite its undeniable shortcomings, the Vandersteen Model II ($880 in our area) is the best speaker system known to us anywhere near the price. The upgrade from the DCM Time Window is clearly audible and worth the $200-plus difference in our opinion. The money will be recouped in some of the other categories below.

Power Amplifier

The sensational Hafler DH-200 ($399.95 wired, $299.95 in kit form) is the hands-down choice here. And it's quite a bit less costly than the Audionics CC-2, our previous and still far from unimpressive nominee. Can't ask for much more, right?

Preamplifer

We offer you two options. Get the *improved* Hegeman HPR/CU ($720), which is in many ways the second best preamp we've found at any price, or get—nothing. The latter option, predicated on the use of the Win Laboratories strain-gauge phono transducer (see below), is unlikely to appeal to those who want a conventional panelful of controls, so we won't push for it too vigorously, although the sonic results are quite possibly superior. But you can't go wrong with the Hegeman, either.

Phono Cartridge and Step-Up Device

If you go the Hegeman route, we recommend the Fidelity Research FR-1 Mk 3F moving-coil cartridge ($230) as before, but this time with a Marcof PPA-1 pre-preamp ($119.95). The Cotter transformer, which we recommended at a time when we weren't aware of any respectable low-cost alternative, has meanwhile priced itself completely out of the Reference B category. The Marcof isn't quite as perfect sonically, but it's truly excellent and in conjunction with the Hegeman yields better results overall than the previous lopsidedly priced preamp/step-up combination.

If you're willing to do without a preamp, get the Win Laboratories SDT-10 Type IIC phono transducer system with the SPG-10 passive volume control module ($550 plus $150). This will save you $370 and even provide you with one switchable "aux" input for a tuner or whatever. We'll vouch for the sound but not for the convenience.

Tone Arm

You may decide to go with an integrated turntable/arm unit (see below), but if you end up with a separate tone arm, we still don't know of a better one at a moderate price than the Series 20 Model PA-1000 ($150).

Turntable

We regret that we can't give you a definitive recommendation in this category as of press time. The Kenwood KD-500 direct-drive turntable, which was our previous choice, is no longer made; several successor models look promising, but we haven't tested them yet. Sony and Yamaha have also come out with some interesting new medium-priced turntables with arms; what's more, adjustable VTA during play is becoming a standard feature on the new Japanese integrated models and may turn out to be the decisive factor in our final recommendation. Dual and Thorens are also in the running with interesting new integrated units.

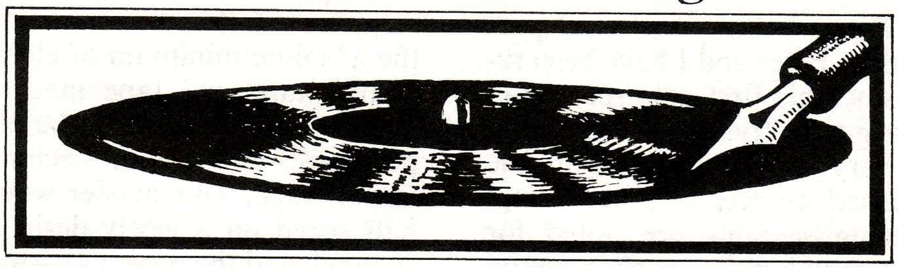
If you've read and assimilated everything we've said on the subject so far, you're in a pretty good position to make your own selection. Look for a base that sounds as dead as possible when you tap it or scrape it; stay away from units that have absolutely no give when you jounce them. The Cotter B-2 isolation platform ($150) is still highly recommended where mechanically transmitted feedback is a threat.

In the next update of Reference B, we expect to have the full benefit of our extensive new turntable tests; we promise to be more specific then.

* * *

No matter how we combine and add up the above recommendations, the highest figure we can get is about $2900; the lowest is well below $2500. We think that's good news considering the greatly improved sound of Reference B.

Records&Recording



***Editor's Note:** This is the Max Wilcox article that barely missed getting into the last issue; the recording sessions it focuses on are now ancient (i.e., 1978) history, but the moral for the big record companies and their producers remains as timely as ever. Note also that the article is actually a sequel to Max's digression, two issues ago, from his original series, which is now beginning to look like his Unfinished Symphony. But then editorial rigidity isn't what you read **The Audio Critic** for.*

Even Less Is Even More

By Max Wilcox

No, I haven't weakened. Those simplified microphone techniques using superior-quality microphones have continued to produce excellent studio results. In this article I'd like to take you behind the scenes of a recent recording project where further refinements were made in the "less is more" approach discussed in Volume 1, Number 5.

For the last few years, Peter Serkin has been devoting his musical energies to Tashi, a chamber music group he formed with violinist Ida Kavafian, clarinetist Richard Stoltzman and cellist Fred Sherry. Peter's last solo recording was a monumental performance of Olivier Messiaen's "Vingt Regards sur l'Enfant Jesus" which he recorded in 1973. Last season Peter gradually resumed his solo career, and his recitals were made up entirely of works of Chopin. For some months he and I worked on plans for RCA recording sessions of this Chopin repertoire, and the sessions were finally scheduled for the middle of July, 1978. In June, Peter and I got together to make final arrangements.

* * *

The first decision to be made was the choice of a recording hall. Since RCA's Studio A was fully booked for the July dates Peter was available, I invited him to visit the hall that had been used for the very first recording I ever produced. The recording was the Brahms F minor Piano Sonata played by Artur Rubinstein, and the hall was the concert hall of the American Academy of Arts and Letters at 155th Street and Broadway in New York City. It is a beautifully resonant hall seating about 800 people and is far inside the building, away from the traffic noise that plagues so many New York recording locations.

Peter went to the hall with me a few weeks before the scheduled sessions and immediately fell in love with the place. It is one of the most beautiful 19th-century style halls anywhere in the world, and since the Academy does not have a concert series, it is rarely used. The decay time of the hall's reverberation is nearly two seconds, and since it is all wood and plaster, the resulting ambience is warm and lustrous. Peter felt it would be ideal for Chopin.

So, the hall was booked. Now we needed a piano. We visited the Steinway basement a few days later, and, after trying many familiar and unfamiliar instruments, Peter chose a CD492, a brand-new piano with a particularly rich, warm tone. The action was still new and stiff, but Peter felt any problems it might cause would be more than compensated for by the beautiful sonority. Steinway and I then arranged for Steve Borell to be the tuner in attendance at the sessions. Steve is one of the half dozen great piano technicians in the world, and we felt he could remove the "newness" kinks from the action during the recording sessions.

* * *

I then presented Peter with my technical proposals for the session. Peter and I have been recording together since his first solo records in 1965, so we are long-time friends and collaborators. We work very closely together at the technical and musical aspects of a recording. Our Tashi recording sessions are noted for their lengthy and painstaking microphone setup tests. Everyone listens very carefully and participates enthusiastically in whatever experiments with microphones and instrumental setups we devise. It's all aimed at achieving a natural and well-balanced sound in the studio before proceeding with a recording session.

Peter is also used to the luxury of multitrack tape recording, with its capability for subtle remixing during the transfer to the final two-track master. The remixing on Tashi records has been minimal, since we devote so much time to achieving the correct balance and instrumental perspective at the session. Still, Peter always felt comfortable knowing we could rebalance a section if some instrument seemed momentarily obscured. His last piano solo recordings had been multitrack, and he had taken advantage of this to change discreetly the microphone balance during various sections of a composition such as the Messiaen "Vingt Regards".

I was therefore a little apprehensive about his reaction to what I was going to propose. I hoped I could convince him of the superior quality these proposals could give the final disc. Since the last Tashi sessions of Mozart and Beethoven Trios had been done with a greatly reduced complement of microphones, I knew Peter was convinced of the scientific and artistic validity of that approach. Still, they were done multitrack, which meant they could be remixed during the final mastering to two-track tape. Since piano solo recording quite obviously does not involve balance with other instruments, it seemed the proper repertoire to begin Phase Two of my new (though for some people very tried and true) technical approach. I therefore made the following proposal to Peter: I wanted to record the piano with two Schoeps/Studer MK2 omnidirectional microphones connected to a custom-made "minimum electronics" console (about the size of a large briefcase), which would be connected to an Ampex ATR-100 two-track tape recorder operating at 30 inches per second using Ampex 456 tape with no Dolby. The object: carry the response of the microphone capsule through the absolute minimum of electronics to a high-quality tape and tape machine. The original session tape would then be edited for musical purposes, and would become the two-track master tape. This master would then be cut at half speed on a newly designed RCA lacquer channel, and the pressings—well, we would face that problem when the time came.

* * *

To my delight Peter's reaction was very enthusiastic, and I began to make the necessary arrangements for equipment. Ray Hall was to be the recording engineer and Ray Rayburn was in charge of the technical setup. As simple as it was, it all needed to be in perfect operating condition to guarantee the full potential of the approach. The day before the session was devoted to draping certain hard surfaces of the American Academy listening room, placing the monitors in proper relation to the console and producer listening area, and tuning the response of the monitor playback system to give the flattest possible playback response at the chosen listening area. This is done by placing a calibration microphone at the chosen spot and adjusting the response of each speaker while feeding it pink noise. The calibration microphone registers the frequency curve of the monitor system and the response of each monitor channel is then equalized throughout the frequency spectrum until the response is as flat as possible at the chosen listening site. *(This doesn't, however, guarantee correct time-domain response, but Rome wasn't built in a day.—Ed.)*

Since the listening room at the Academy has a large bass peak all too familiar from my Rubinstein sessions in days gone by, the monitor tuning made a reasonable control room out of a room that is otherwise so weighted at the bass end that critical evaluations are quite impossible. (This was the reason we discontinued the Rubinstein recordings there in the early 1960's. The resulting records were fine, but the listening was very difficult, and monitor tuning was not a well-known art in those days.)

Our first Serkin session was scheduled to begin at noon. That gave us the morning for last-minute technical adjustments and gave Steve Borell ample time to begin the fine-trim on the action of the CD 492. Peter walked in at 12 o'clock and suggested that we should all have lunch while he would get used to the piano. He then went on stage and, since these were Chopin sessions, started playing a Bach Invention of course. As soon as he started to play I asked Ray Rayburn to push the record button. Since he thought we had left, Peter played for a few minutes and then walked back into the control room. I said, "Would you like to hear that?" After letting him recover from his surprise that his Bach had had an audience, I seated him at the calibrated listening position and played back the Bach Invention. After about a minute of listening he said, "That's the best recorded piano sound I've ever heard. Don't change anything!" I then joked that this was a world record for us in briefness of microphone tests and went buoyantly off to a Westphalian ham sandwich.

Peter happily recorded for five whole days, and Ray Rayburn changed tapes like mad (since at 30 IPS you can record a total of only 16 minutes per reel). We never moved a microphone an inch because the sound seemed to fit all the repertoire, and at the end Peter said it was the best series of recording sessions he'd ever had.

* * *

Now, of course, there is absolutely nothing revolutionary about this approach. Gordon Parry and John Culshaw recorded Solti's entire historic Wagner "Ring" cycle on two-track tape, and it is still general practice at English Decca (London Records). Many quality-conscious independent recording engineers have been using this approach for years. Why did the major companies gradually desert two-track recording? Well, let me give a brief history of the development of the number of tracks on tape machines.

Until about 1956, all tape machines were two-track and used quarter-inch tape at 15 or 30 IPS. Then the three-track machine, using half-inch tape, was developed. This meant the session tapes could be edited, and then remixed to the final two-track master. Various instruments, like the soloist in a concerto or the woodwinds in an orchestra, could be put on the center track, and the balance could be altered in the final mix. It also began the subtle but real degradation of the quality of the original tape. Besides going through the electronics of the console of the original session, the tape was then played back through a remix console and copied to a two-track tape. Each time a tape is copied the crispness of the transients and the general cleanness of the sound are slightly compromised. Nothing drastic, but it's not quite as good as the original. Three-track tape also began the use of more microphones. If we have more tracks, we can use more microphones, right? Artists quickly learned that the balances they heard at the session could be changed in the mix. Conductors were often surprised at the ultimate balance of a concerto recording when the record was issued. All was well balanced at the session, but when the record was issued, the soloist had often grown to larger-than-life proportions. Soloists would often gently but firmly influence their producers to alter the final mix in their favor. And so it all began. Four-track, eight-track, sixteen-track, twenty-four-track. More and more microphones fed bigger and more complex consoles with more electronics between the microphones and the tape.

As more tracks permitted more microphones, the concept of an overall sound on an instrumental group faded away. It was replaced by separate microphone setups on each instrument or group of instruments. Since these separate pickups were meant to control the balance of each group, the microphones were almost always used in their cardioid (directional) configuration. Cardioid microphones give lots of control. They also give a rather peaky, beamy sound with a large roll-off in the bass beginning at 50 Hz. What they do give is lots of *control*. Multitrack tape machines and cardioid microphones have made *control* the most important word in the recording studio. Pop records are usually based on musical arrangements that were never intended to be heard balancing themselves in a natural acoustical environment. They are written for instruments separated by acoustical panels playing in a dead room into microphones a few inches away from the sound source. It is all mixed, equalized and echoed both then and later, and it makes a lot of money for a lot of people.

The trouble begins when you apply this approach to classical music. As multitrack recording began to dominate the classical recording scene, the philosophy of where a recording was actually created began to change.

Many engineers and producers began to consider the actual session as no more than the beginning of a long process. It became "Get the sound on the tracks and we'll balance and equalize it in the mixing room." No longer was a conductor really in charge of creating and approving the sound and balance of his orchestra at a session. The producers could say that they would correct any balance problems in the mixing room. All very convenient and economical, but the artistic control was taking a subtle and questionable shift. If a horn solo is covered in a recording, it is usually because the accompanying strings are playing too loud. The correct solution would be to ask the conductor to replay the passage with the proper balance. That would produce a natural sound that could not really be achieved by lowering the string tracks and raising the horn in the remix. It would also be subject to the musical taste of the conductor, not that of the producer.

The actual sound of the ensemble is also not sacred at multitrack sessions. What the conductor hears in the playback room at the session is usually a flat playback of the tape. When the multitrack tape is remixed by the producers, many of them add dramatic equalization changes which quite alter the texture and sonority of the ensemble. If the strings didn't sound bright enough at the session, wouldn't it seem more logical to change microphones and microphone positions until the conductor and producer were both happy with a result that would then be preserved until the final record?

* * *

In far too many cases, the multitrack, multimike world has become a place where producers and engineers treat the players and conductors as tools to be used as they create "their" performance and balance. Listen to Giulini's recording of "Pictures at an Exhibition" with the Chicago Symphony on DGG and hear the percussion section move forward and back, like a dancer on a runway, as the music progresses. Did the producer and engineer know more about orchestration and orchestral balance than Maurice Ravel? Did the Chicago Symphony sound like that when they played at the session? It's all contrived to "wow" the listener as he turns on the hi-fi rig and is knocked out by the impact of the cymbals. Ravel's masterly orchestration of Moussorgsky's music becomes a playground for electronic razzle-dazzle. Of course, the most dazzling thing of all would be a properly balanced, full-dynamic-range, honestly recorded performance of what actually goes on during a good performance of "Pictures at an Exhibition".

Thank goodness the pendulum is swinging back. The current audiophile recording scene is based on a return to the reality of the actual sound of music. Superb electronics capturing the sound of a few, properly placed, calibration-flat microphones is infinitely more exciting and, yes, more "commercial" than all of the tortuously remixed, re-echoed, re-equalized constructions that emerge from multitrack mixing rooms. When you hear a great symphony orchestra in Carnegie Hall, do you wish you could put in a 6 dB boost at 8 kHz so the strings would sound more brilliant? Well, neither do I, so why should it be done on a record? Just because a producer likes "bright" string sound?

We will look back on all of this as the Dark Ages of electronic meddling, and I'm quite confident that natural sound will prevail once again. Does the true sound of a Guarnerius violin need improving? What it presents is a challenge. To capture the actual sound of such an instrument requires precision recording equipment used in a beautiful acoustical environment. Equalized cardioid microphones will never capture that sound, and neither will the people who operate them. "Control" of the sound has never been the object of true audiophile recording. A rock band playing parts that were never designed to balance each other without electronic help needs "control." Maurice Ravel's orchestration was designed to work. It needs to be captured.

A whole new generation of recording people are going back to the basics and coming up with superb results. Lincoln Mayorga and Doug Sax returned to the basic principles of 78 RPM disc recording when they initiated the "direct to disc" techniques at Sheffield Labs. That began the whole current resurgence of audiophile recording, and the classical music record collector will be the ultimate beneficiary. Flat, natural-sounding equipment needs flat, natural-sounding program sources to demonstrate what it really can reproduce. Lots of us are trying to give it to you. The independent labels are at the head of the pack at the moment. Let's hope the majors, with their great artists and resources, will not be far behind.

A Discography for the Audio Purist: Part II

Our ground rules for this series of record reviews were explained briefly in the last issue; we left out, however, one important point. It concerns the new category of "audiophile" or "super high fidelity" records, often though not invariably direct-to-disc, sold at a huge premium and marketed much the same way as esoteric audio components.

We wish to make it clear that such records should under no circumstances be assumed without prior knowledge to be superior to the regular releases of the major commercial labels, especially to the European imports (Deutsche Grammophon, EMI, Philips). The audiophile jobs are, as a class, amateurish in musical performance, microphone placement and general producer savvy, even when their tape and lacquer channels are exceptionally clean—which they aren't always. There are some notable exceptions and we certainly intend to single them out here, but on the whole the Mitch Cotter or Dick Sequerra type of boutique-made audio component doesn't have its qualitative counterpart in phonograph records; it just doesn't seem to work out that way.

One particular branch of the premium-priced, super-audiophile record market has us especially worried, as it has been hailed the wave of the future and doesn't live up to that billing in our opinion. We're talking about the new digitally recorded discs, such as for example the recent releases on **Telarc** (distributed by Audio-Technica). The digital recording process used to generate the master tapes for these records has a sampling rate of only 50,000 samples per second, which appears to be the currently accepted unofficial standard but lacks the high-frequency resolution of state-of-the-art analog recording. We must admit that the bass definition and dynamic range of the digital recordings are phenomenal, but we hear some unmistakable top-end degradation through our Reference A system. We estimate that 100,000 to 120,000 samples per second with 18-bit encoding/decoding would realize the ultimate sonic potential of digital recording, at which point analog techniques would definitely be put in the shade. There's not a thing wrong with digital recording as a concept, but you can't do it by halves. We'll have a lot more to say on the subject in future issues.

Meanwhile, we want to tell you about just a handful of recently issued discs of "ultimate" quality, without resurrecting this time any of our older favorites. There will be plenty of opportunity for that as this series continues; this particular issue is much too crowded to leave room for anything but the choicest items of current interest.

Proprius

Our small collection of reference-quality records is getting top-heavy with the Proprius Böcker & Musik (books and music) label; it just so happens that this little-known Swedish company continues to manifest higher and more consistent standards, both technically and musically, than any other record maker we can think of. (See also our comments on this in the last issue.) It seems that every time we want to eliminate the program material as a potential source of sonic obfuscation in a A-B listening test, we reach for a Proprius record. Enough said.

* * *

Dalakoraler och Brollopsmusik (chorales and wedding music from the Dalarna region of Sweden). Bengt Granstam, organ, on the two Magnusson instruments of the Stora Tuna church. Proprius PROP 7763 (made in 1976).

This is just about the most real-sounding organ record known to us, utterly transparent and totally delineated. Just listen to those 8-foot Spanish trumpets *en chamada.* Whew! And the bass from those 16-foot stops goes *all* the way down, without distortion. Add to that the excellent, highly musical playing of Bengt Granstam and all that's missing is J.S. Bach. But the Dalarna church music is yoost fine for pleasant listening.

* * *

Jazz at the Pawnshop (recorded live in December 1976 at Stampen in Stockholm). Arne Domnerus, alto sax and clarinet; Bengt Hallberg, piano; Georg Riedel, bass; Egil Johansen, drums; Lars Erstrand, vibraphone. Proprius PROP 7778-79 (two-record set).

If we were allowed to name only a single example of what we consider to be flawless, natural-sounding recording, this would be the one. It sounds like a jazz quintet in a night club, period. You're sitting at one of the tables and they're right there, before your very eyes and ears. Most of the latest crop of direct-to-disc jazz records sound utterly phony and crudded-up next to this elegantly taped production.

The fact that the jazz played here is strictly mainstream and relatively tame ("Lady Be Good" and suchlike) is rather beside the point. Excellence in recording and musical originality have seldom gone hand in hand in jazz. The playing of these five Swedes is quite slick, smooth and expert, in any event; the results are thoroughly stylish and musical, even if not exciting. Get this album just as a standard of comparison.

* * *

Laudate! (Sacred music of the 1600's from the Uppsala University library collection.) Uppsala Academic Chamber Choir; Drottningholm Baroque Ensemble; Anders Eby, conductor. Proprius PROP 7800 (made in 1978).

This is an even better choral record with soloists than the *Cantate Domino* we reviewed last time or the *Kör* we also mentioned. Better in sound, that is; in musical interest it's a bit more specialized. But there's no tape hiss or modulation noise; the cut is cleaner; the texture is even more transparent—and on top of it the chorus is better, with more secure intonation. That makes the record just about State of the Art for this sort of thing. It couldn't sound more natural. Obviously, the Proprius people keep improving their already formidable technique.

Reference Recordings

First Takes. Andrei Kitaev, piano; Bill Douglass, acoustic bass. Reference Recordings, Jazz Series, RR-6 (45 RPM, made in 1978).

A typical super-audiophile label if there ever was one, RR never particularly impressed us with their older "Limited Edition, Classic Series." This new Jazz Series release seems to take another tack, which includes 45 RPM and a different recording engineer. The results are spectacular, to put it mildly.

This is the best jazz piano sound we've ever heard off a piece of vinyl and one of the few piano records regardless of musical content that we find truly clean. It may have something to do with the Grotrian Steinweg Imperial concert grand, but the impact, dynamics and delineation of the piano are simply stunning. No shattering ever, from the first groove to the last. The acoustic bass is also beautifully recorded; both the fingernail transients and the lowest fundamentals are audible and clean at all times. A magnificent job.

Musically the record is also interesting. Andrei Kitaev, a conservatory-trained Russian in his late 20's, cites Oscar Peterson as one of his influences, and his music-making reflects it. This is no meatball, Eastern-bloc imitation of contemporary American jazz. It's talented, idiomatic and exuberantly musical improvisation in the best jazz tradition. The excellent American bassist is a big help, of course. What's really remarkable, though, is that this was supposed to be a *test take* for a later recording session, by two musicians who had just barely met each other. The tape rolled for an hour and this was the result. That's jazz, baby.

—Ed.

Subscription Information and Rates

First of all, you don't absolutely need one of our regular subscription blanks. If you wish, simply write your name and address as legibly as possible on any piece of paper. Preferably print or type. Enclose with payment. That's all.

Secondly, we will not send **The Audio Critic** any other way except by first-class mail in the U.S. or Canada and by airmail overseas. We simply refuse to be blamed and hated for the vagaries of postal delivery by cheaper methods. You're the one who pays for it this way, it's true, but you'll get much more reliable service.

That means we have only two subscription rates. If you live in the U.S. or Canada, you pay $30 for a subscription of six consecutive issues by first-class mail. If you live in any other country, you pay $36 for a six-issue subscription by airmail. All payments from abroad, including Canada, must be in U.S. funds, collectable in the U.S. without a service charge.

You may start your subscription with any issue you desire, as long as we still have it in print. You'll get six issues for your money, no matter when and where you start. Right now our available supplies go back only as far as Volume 1, Number 4 (the issue that includes the famous cartridge/arm/turntable alignment article). If you don't specify which issue you want to start with, we'll assume it's Volume 1, Number 6, our cumulative reference issue with over 150 reviews.

One more thing. Please don't write us asking for a fractional subscription, whether it's for a single issue, three issues or whatever. The answer will always be no. Our rule is a full six-issue subscription or nothing. You're fully protected, however, by our policy of refunding the unused portion of a canceled subscription. And once you're a subscriber, you may order extra copies of any issue in print at $5 each. In addition, individual copies are available to all comers at $6 each in selected audio stores. But not from us.

Address all subscriptions to The Audio Critic, Box 392, Bronxville, New York 10708.

The Audio Critic

In the next issue:

We conclude the seminar on the State of the Art with Part II of the transcript. Whew!

We report our long-delayed comparative tests of mechanical resonances and acoustical breakthrough in turntables and tone arms.

The factual and mythological aspects of speaker wires and audio cables are examined in detail.

Our "benign neglect" of FM tuners comes to a reluctant end, one issue later than promised.

We take our first critical look at tape recording and tape decks.

Also, other equipment reviews in all categories, as well as the usual features and columns.